DESIGN OF AN AUTOMATIC COMMUNICATIONS SWITCHING SYSTEM

DUDLEY N. KYLE

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Submitted in partial fulfillment of
the requirements for the degree of

MASTER OF SCIENCE
IN
ENGINEERING ELECTRONICS

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ABSTRACT

The time lag in conventional voice communication circuits involved in the use of new techniques of warfare cannot be tolerated. This time lag has led to an ever increasing demand for more voice channels, where in reality what is required is a more efficient use of present channels by reducing the time lag. To answer this requirement a design of an automatic communications switching system is proposed. The system is designed to meet the voice communication environment created by military tactical data systems such as the Marine Tactical Data System. In addition it is intended as a proposed answer to the voice communication requirements of military headquarters such as battalion and higher. Using this system an operator can rapidly and automatically select any desired voice channel to enable voice communication with any other operator in the system with a minimum of time delay.

The writer wishes to express his appreciation for the assistance given by Mr. Sidney Hasin of Litton Industries, Beverly Hills, California, without whose help this investigation would have been much more difficult. In addition, thanks are due Professors Mitchell L. Cotton and Donald A. Stentz for their valuable advice and encouragement.

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TABLE OF SYMBOLS AND ABBREVIATIONS

A–C	Alternating Current
D–C	Direct Current
d–c	Direct Coupled
F.F.	Flip-Flop - Binary Multivibration
Helihut	Helicopter Transportable Hut
H.F.	High Frequency
M.T.D.S.	Marine Tactical Data System
T.A.C.C.	Tactical Air Control Center
T.A.O.C.	Tactical Air Operations Center
U.H.F.	Ultra High Frequency
C	Capacity
E	Volts
e	Instantaneous Volts
db	Decibel
f	Frequency
h	Height
J	$\sqrt{-1}$
l	Length
λ	Conductivity
M	Mutial Inductance
n	Number of Stations
r	Number of Connections at a Time
ω	$2\pi f$
Z	Impedance
	"And" Gate
	"Or" Gate
	"Inhibit" or "Not" Gate

1. Introduction.

In every military command there are several types of communications channels available within the command. However, the various channels are seldom, if ever, so interconnected that there is common access to all the channels that exist. The usual case finds a wire and radio relay net terminated in switchboards to provide local and long distance telephone and teletype communication and several radio nets terminated in a message center of some form. In addition there are usually various operational radio nets terminated at various locations throughout the command to provide specialized communication for various supporting functions. The time lag inherent in this type of setup is obvious; for instance, to place a telephone call a switchboard operator must be contacted and instructed, then a wait ensues while he then sets up the call before conversation may begin.

The time lag is even worse when information must pass over a radio net, for now the information must be written down, transmitted to the message center where, according to its priority and the number of messages ahead of it, it will await its turn to be transmitted. In many situations this time lag cannot be tolerated, which has led to the establishment of private or "Hot" channels of communication whereby two or more parties may converse without the time delay involved in using the normal communications available. This, of course, leads to considerable inefficiency in the use of equipment and information flow between commands. For instance, when the normal channels are overloaded there is seldom a way the private channels, which normally will be working far below capacity, can be utilized to handle the overflow.

Perhaps the outstanding example of where the problem of time lag in

communications must be overcome is in Air Defense. Here, with modern supersonic aircraft, missiles, and nuclear weapons, decisions must be made rapidly and efficiently. These decisions must be communicated rapidly to the personnel who will carry them out and any time delay in obtaining and using available communications to transmit the decisions may mean the difference between success and failure in preventing the delivery of a large area destruction weapon. Recognition of this threat inherent in the advent of high speed aircraft and missiles armed with nuclear weapons has led to a study of the present means of air defense to determine what improvements must be made to counter this threat. This study has led to the award of a contract[1] to Litton Industries, Beverly Hills, California, for three phases of work as specified in the Contract Specification[2] for a Marine Corps Tactical Data System. Initial results of this work[3] indicate that the effective functioning of such a system, and the mechanization of its communications are of a necessity interlinked.

As may be seen from figures 1 and 2, MTDS employs a decentralized mode of operation, thus imposing strict requirements on the communications network. For in this type of operation where the operators are in relatively isolated helihuts, it is communications that is the nerve fiber of the system. It is via the communications system that situations are discussed, decisions arrived at, and orders transmitted. Any delays encountered by operators in obtaining access to one or more other operators may seriously degrade the ability of MTDS to react to a given situation. Thus the ability of the system to effectively counter enemy threats will to a large extent depend upon the speed and ability of the operator to utilize the communications provided. Therefore, the problem that exists in MTDS is the same problem that exists in every command—that of interconnecting all the communica-


AIR
CENTER


TACTICAL AIR
CONTROL CENTER

tions provided and presenting them to all personnel so that they may use them in the most rapid and efficient manner possible.

From the above examples it is readily apparent that a definite requirement exists for a communications switching system that will make possible the rapid, efficient use of all available communications channels. Since MTDS imposes such strict requirements upon such a switching network, it has been chosen as the system which the network must satisfy, for if the system satisfies MTDS requirements it may also be utilized in many other less demanding situations such as unit command posts.

2. Study of System Requirements.

A necessary first step in the design of any system is determining what requirements the system should satisfy. Accordingly, an operational study of MTDS was undertaken to determine the requirements it would demand of such a switching network. In addition, the possible application of the switching network to other situations such as battalion or higher headquarters were considered which generated other requirements.

An early result of the analysis was to reveal that the system may be subdivided into two major parts, (a) an operational system that interconnects all the personnel concerned with the actual operation of the system or unit, and (b) an administrative system which interconnects all personnel concerned with the maintenance and administration of the system or unit. This separation is desirable for much stricter requirements are placed upon the system by the operational personnel, especially with regard to the reduction of the time lag inherent in normal communication. Administrative and maintenance personnel, however, are usually quite adequately served by normal communication systems, hence require a far simpler switching network.

When all the requirements generated by MTDS and the other units are considered, they are remarkably similar as may well be expected. For, as pointed out earlier, the problems faced by MTDS and by other units in efficient use of communications are much the same; the real difference being MTDS requirements are stricter and more pressing than the others. Considering all factors the requirements for the operational communication system may be stated as follows:

a. Each station must be able to selectively call and establish a voice channel with any other station in the system by a simple operation, such as

pushing a button or buttons.

b. There must be no busy signals, that is, any station must be able to call any other station regardless of the fact that the called station is busy on another channel.

c. The called station must have a visual indication of who is calling him.

d. The voice channel must not be established until the called station accepts or answers the call.

e. For operation with systems that may utilize a computer it is necessary to provide an output that indicates when a voice channel has been established between two or more stations in the system.

f. A station must be able to connect (or party) as many other stations on its line as desired, for conference calls.

g. A station must be able to monitor any desired voice channel in the system while maintaining the capability to talk on other channels.

h. Each station must have access to external communication channels through the communications switching network or via a manual switchboard.

i. A station must be able to select a UHF receiver-transmitter, and any one of 21 channels on the selected receiver-transmitter.

j. The station should be provided with a dual headset and microphone. One earphone is associated with the microphone while the second earphone is used to monitor.

k. The system should provide a minimum capability for communication channels as follows:

1. 20 internal stations
2. 15 external trunks
3. 8 air ground radios (UHF)

1. It should be readily expandable to provide more stations, if desired.

In the above list of requirements an internal communication channel is defined as a channel connecting operators within an operational area such as a Command Post. An external channel is defined as a communication channel which connects two or more operational areas.

Since many of these requirements are for the most part the result of compromise decision, they will be discussed in turn, in greater detail.

1. Requirement a. is obvious as this is the prime purpose of such a system.

2. It was felt that modern situations where such fast action and reactions are required of the personnel utilizing the system that no call should be prevented from reaching the intended party due to lack of a channel. It should be the function of the called party, not the system, to accept or deny a call. Further, in many situations, as previously pointed out, the time lag involved in waiting for an idle channel to place a call may be the difference between success and failure in countering a threat. In addition, it is the objective of the system to reduce time lags to a minimum. Thus the reason for requirement b.

3. Since we have established the premise in requirement b. that acceptance or denial of a call is placed upon the called station, there must be provided a means of notifying the called station who is calling. This enables the station to decide how and when to accept the call. Further, the visual indication may be utilized to provide a record of who the station is connected to at all times.

4. Requirement d. is necessary to prevent "break-in" type operation. This means that if a called station is busy it may not be interrupted until

the station desires.

5. The advent of large capacity general purpose digital computers and modern data processing equipment means that much of today's communication will be via data links. Frequently it will be desired that when two stations are in contact with each other they will desire to pass data from a computer between them via associated data links. The computer must be notified which two stations are in contact with each other, hence the need for requirement e.

6. Requirement f. answers the need for several people to talk to each other to resolve mutual problems.

7. Requirement g. provides the capability of continuous listening where required.

8. In requirement h. it is anticipated that not all channels to remote locations will be terminated in the switching network, but will be terminated in a panel or switchboard to enable personnel without access to the switching network to utilize the channels. Further, this allows the stations to have access to additional channels when needed.

9. This requirement answers the need for air ground communication so necessary in this air age.

10. Requirement j. provides the necessary input-output device to fulfill requirement g.

11. Requirements k. and l. are design objectives arrived at after careful consideration of what personnel would normally use such a switching network. The external and UHF requirements are based on what normally would be required for communication between widely dispersed units and their air support.

There is an additional requirement which should be considered in a

system design of this nature: the dual requirement of reliability and simplicity. These two go hand in hand, for it is logical that a simple circuit can be inherently more reliable than a complex circuit. For a system to answer the need of this system, that of enabling people to talk to people, it must be reliable. A failure in the system may seriously prejudice the operation of the unit using it. Therefore, every effort will be made to keep all circuitry simple and to utilize reliable techniques.

3. System Design Objectives.

Now that the requirements for the system have been generated they must be studied to determine what they actually require of the system. Initial study reveals that three basic things must be done:

a. there must be a form of switching to accomplish any desired connection of two or more communication paths.

b. there must be a means of accomplishing signaling of some nature to inform the called station exactly who it is that is calling him.

c. the stations must have a visual record of which station it is in contact with, since the station may be in communication with more than one station at a time.

There are three types of communications switching to consider. Each type has its own unique problem to be considered, yet they each have problems in common. These three types and their problems are:

a. Internal Communications Switching

1. Establishment of desired communications channel.

2. Signaling of called station and identification of calling station at the called station.

3. An output indicating that the communication channel has been established.

b. External Communications Switching

1. Establishment of desired communications channel.

2. Signaling of called station and identification of calling station at the called station.

3. An output indicating a communication channel has been established.

4. Provision for "press-to-talk" on external channels that are via H.F. radio.

c. Air-Ground (UHF radio selection)

1. Establishment of channel to desired receiver transmitter.

2. Provision of "press-to-talk".

3. Selection of one of twenty-one desired channels on the receiver transmitter.

Now that it has been established what the system must do, it remains to be seen how the system will do it. An obvious approach is to just run wires from each station to every other station and do the necessary switching directly at the stations concerned. However, examining this situation by means of the combinational formula

$$^{n}C_{r} = \frac{n!}{r!\,(n-r!)} \qquad \begin{array}{l} n = \text{number of stations} \\ r = \text{number of connections at a time} \end{array}$$

for just the switching of the internal communication channels reveals that for 20 stations connected in all possible combinations two at a time:

$$^{20}C_{2} = \frac{20!}{2!\,(20-2)!} = \frac{20!}{2!\,18!}$$

$$= 190 \text{ connections or wires}$$

It is apparent that this would be entirely too much wiring and the connection for external channels and UHF radio have yet to be considered.

To avoid this multiplicity of wiring it is clear that some form of centralized swithcing is required. In a centralized switching scheme all communications channels are brought into a common location. The station demanding a connection then has its communication channel switched within the central point to the desired channel. In any form of centralized switching, of which a telephone exchange is an excellent example, there

are three essential categories of circuits:

a. the switching circuit proper or speech circuit which functions to bring about any desired connection of calling station to called station. This function transmits the speech and may not contribute to the signaling.

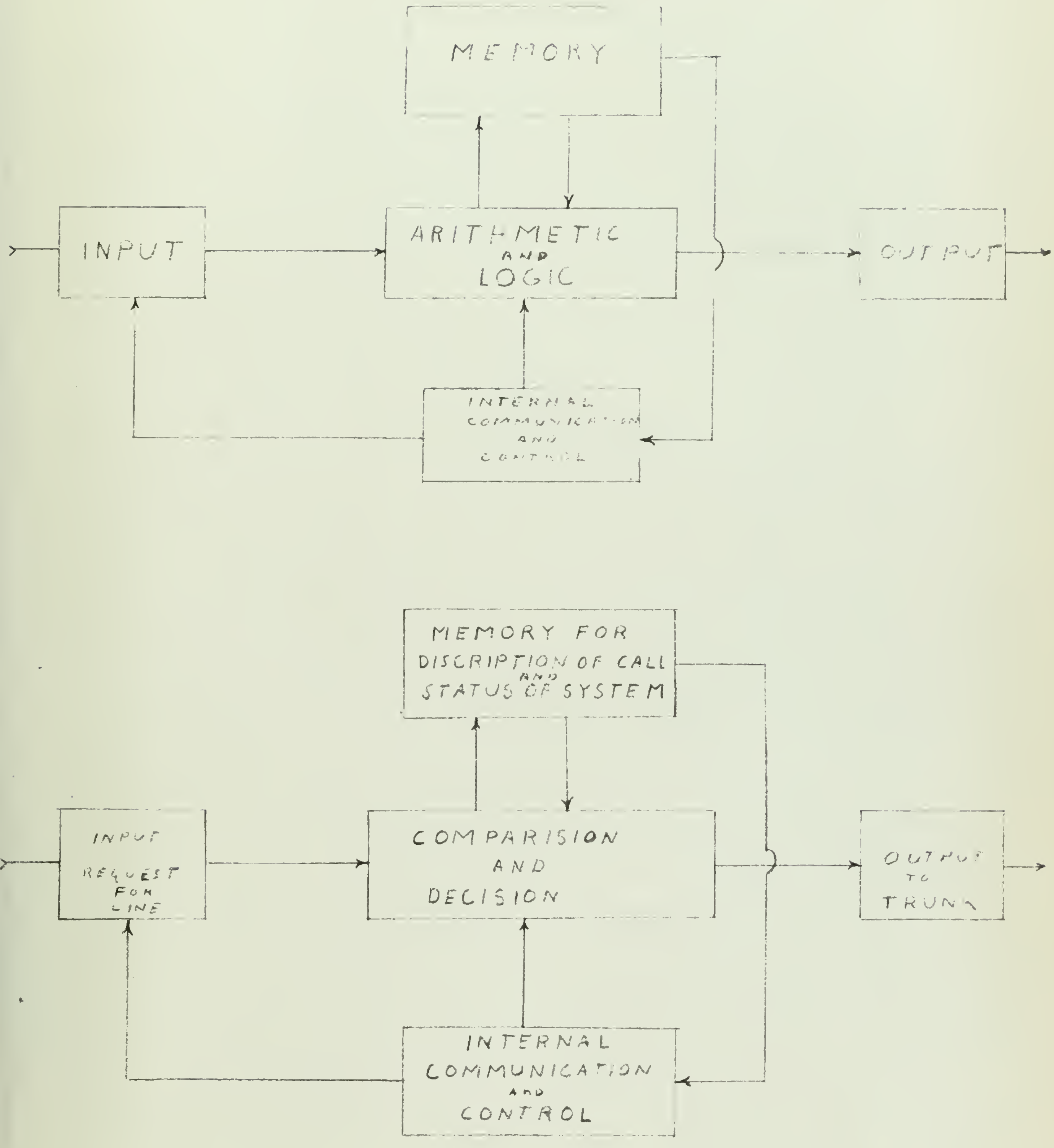
b. the memory circuit whose main role is to receive orders issued by the calling station.

c. the control circuit whose task it is to execute the orders received.

When these three functions are considered it is seen that a centralized switching unit is remarkably similar to a digital computer. A block diagram of these functions when compared with a block diagram of a digital computer, figure 3, reveals that they are indeed alike, differing only in the fact that the switching computer does no actual computation but determines what two channels to connect together on the basis of the information given it.

This close analogy with a computer plus the rapid rise of new forms of electronic components and circuits generated by these computers, have led telephone companies into extensive investigation of electronics switching based on the use of large capacity special purpose digital computers[4,5,6,7]. These systems are in general much too large and complex for their techniques to be used in this system, but the basic idea may be used, that of basing the design on the use of a very special purpose digital computer to direct the switching. While it is not proposed to design a digital computer for a system on this scale, the principles of logical design inherent in the design of a computer and the functioning involved can very well be applied to this problem of communications switching.

FIG 3



Block Diagram of Digital Computer and Automatic Switching System

Having established the fact that the design of the switching system is analogous to the design of a real time computer it now remains to establish a system design.

Before proceeding further the premise is established that it is of no concern in establishing a design, whether the communication channel brought into the switching central is internal, external or UHF radio.

To connect two channels together the switching central requires the following items of information:

a. Identity of the calling station.

b. Identity of the called station.

c. Is a speech or monitor connection desired.

These same items of information are required to accomplish the signaling function desired within the system.

The most efficient way to send this information to the central is in the form of a code. When the various coding schemes are considered it may be shown[8] that a binary coding scheme is a logical choice since it offers economy in storage space and is the simplest coding scheme to mechanize. Thus a binary coding scheme is chosen for use in this system.

For a station to request a connection it must tell the central which of the 42 channels it desires connection to. To send this information in a binary code a six digit code is required since $2^6 = 64$ while $2^5 = 32$. In addition, the station must inform the central whether a speech or monitor connection is required which will require one additional bit of information. Thus a total of seven binary digits or bits must be sent to the central by the station to enable a connection to be made. The central must also be informed as to what station of the 43 possible stations is demanding service. As shown above, this requires an additional six bits of

information. This now requires that a station send a minimum of 13 bits of information to the central to enable the central to make a connection and signal the called station.

It is desirable for the central to operate on a "one at a time" basis. to prevent it from having to operate on two incoming signals at the same time. This situation must be avoided for the following reason:

Assume station A sent a coded signal 000101 and station B sent a coded signal 001010. If both signals arrived simultaneously the central might see this as 001111 which action neither station desires. Thus a means for examining requests for service "one at a time" is desired at the central. In normal telephone practice the speech channel itself is utilized to send the coded request for service to the switching central or exchange. In this system this method of operation is not too desirable for it implies the use of additional circuitry such as line relays and line scanners to determine when and what kind of service is required. In this system it is easier and desirable to use additional wiring rather than circuitry to accomplish desired functions, provided of course that an excessive amount of wiring is not involved.

Thus it appears that a separate code trunk for transmitting control and signal information is desirable. The use of such a trunk has certain advantages in that each station connected to the trunk could be allotted a time interval during and only during which the station could send information to the central. This scheme would mean that the station would have to send only seven bits of information to the central. The central, by keeping track of which station's time slot is present at a particular time interval, would supply the additional information required: the identity of the calling station.

The solution for the control and signaling problem for the internal stations is to run a common code trunk linking all stations in parallel and utilize time division on the trunk. That is, each station is allotted a time slot on the trunk during which the station may send information to the central. This is the solution adopted and is shown in block diagram form in figure 4.

In summary this is the system design philosophy adopted:

a. All speech channels--internal, external and UHF-- are brought to a central point for interconnection.

b. A switching central is utilized to perform the desired connections.

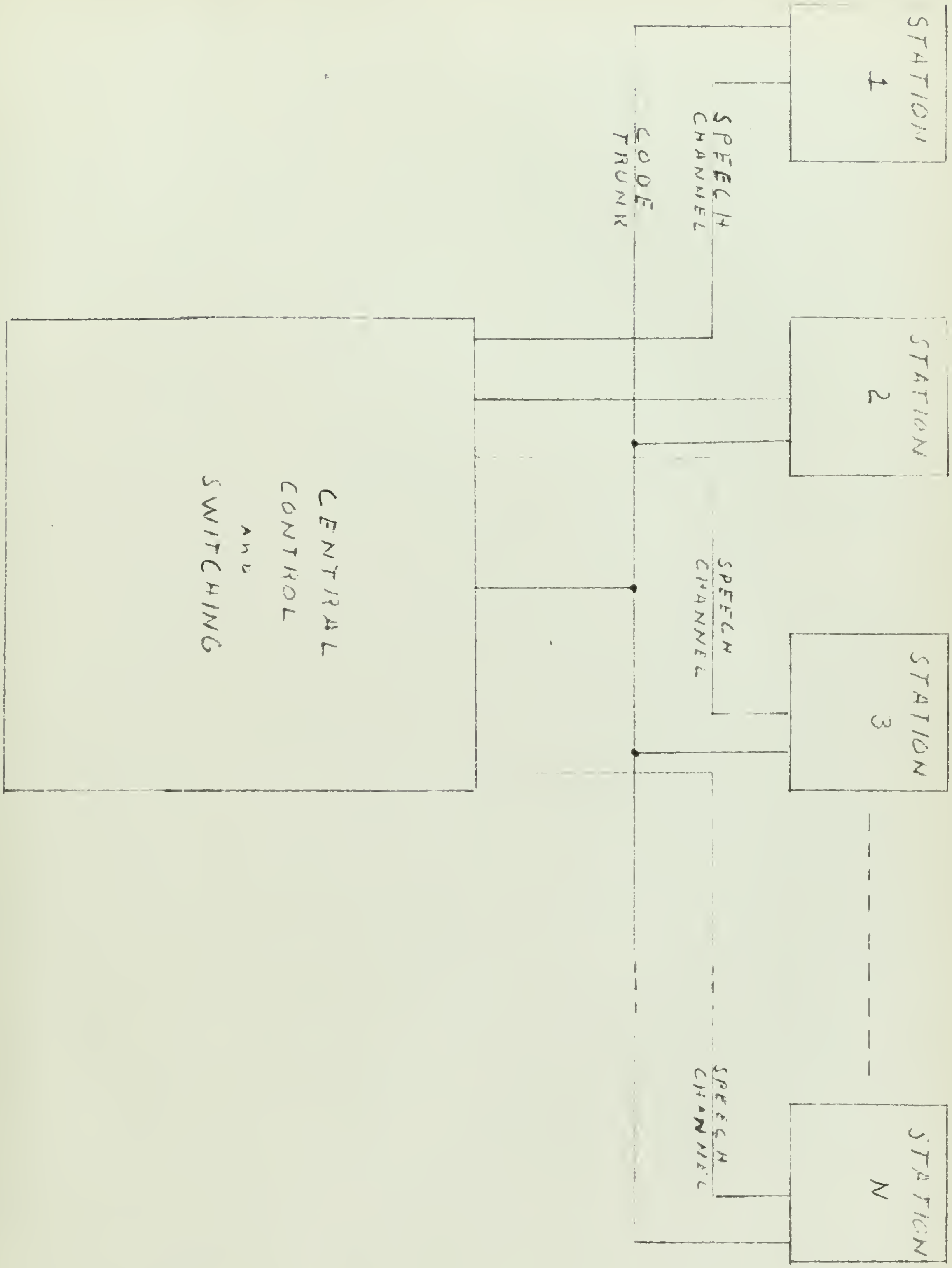
c. The switching central is controlled by binary coded information supplied by the stations connected to the central and by a counter which controls the time slots on the code trunk.

d. This information is sent to the central on a common code trunk linking all stations. The stations time share the code trunk so that only one station may use the trunk during a time interval.

Having decided upon the above switching circuit philosophy, other problems must be resolved before proceeding to a mechanization of the system.

First the problem of what kind of code to generate must be resolved, that is, should the code be generated and handled in serial stream or parallel. The serial stream has the advantage of requiring the minimum of only one wire to carry the code, but has the disadvantage of being complex to generate at the input device and requires higher speed in the processing logic circuitry to handle it. The parallel mode has the disadvantage of requiring a wire for every bit, but has the advantages of:

FIG 4



Block Diagram of Switching System

a. higher speed of operation.

b. use of direct-coupled logic.

The use of direct-coupled logic, that is, signals represented by steady-state voltage, avoids a design problem inherent in A-C circuitry. In A-C circuitry to obtain the "and" function precise coincidence is required of the pulses to be applied to the "and" gate. Since delays are encountered in any system as the pulses progress through the logic, the design of circuitry to cause this exact coincidence at an "and" gate can be quite complex. The A-C system also lacks some flexibility since the addition of any function generally requires the overhauling of the logic circuitry to maintain the proper signal coordination. With D-C circuits unequal chains of logic may exist with no particular problems.

It is apparent that the two methods would be combined, that is, the code be generated in parallel and sent in serial stream which would be a saving in wire. However, additional circuitry would be involved in the conversion at each end of the trunk from serial stream to parallel. As stated before, it is desirable to utilize wiring rather than circuitry provided the cost of additional wiring is not too great. Therefore in view of all these factors, the D-C parallel mode of operation is chosen.

It should be noted before proceeding further that the use of a six digit code allows expansion of the number of stations the system may now handle. Examination of a six digit code table (figure 5) reveals that:

a. The code table may easily be subdivided into the three parts shown.

b. This subdivision allows the sixth bit to be used to identify a channel as an internal or external channel, thus the station that channel represents to be identified as an internal or external station.

FIG 5

BINARY CODE TABLE

BIT NUMBER

	6	5	4	3	2	1
1	0	0	0	0	0	0
[illegible]	0	0	0	0	0	1
	0	0	0	0	1	0
4	0	0	0	0	1	1
[illegible]	0	0	0	1	0	0
[illegible]	0	0	0	1	0	1
7	0	0	0	1	1	0
[illegible]	0	0	0	1	1	1
9	0	0	1	0	0	0
[illegible]	0	0	1	0	0	1
	0	0	1	0	1	0
[illegible]	0	0	1	0	1	1
[illegible]	0	0	1	1	0	0
[illegible]	0	0	1	1	0	1
	0	0	1	1	1	0
16	0	0	1	1	1	1
[illegible]	0	1	0	0	0	0
[illegible]	0	1	0	0	0	1
[illegible]	0	1	0	0	1	0
[illegible]	0	1	0	0	1	1
[illegible]	0	1	0	[illegible]	0	0
	0	1	0	1	0	1
	0	1	0	1	1	0
	0	1	0	1	1	1
[illegible]	0	1	1	0	0	0
[illegible]	0	1	1	0	0	1
[illegible]	0	1	1	0	1	
[illegible]	0	1	1	0	1	1
[illegible]	0	1	1	1	0	
	0	1	1	1	0	1
[illegible]	0	1	1	1	1	0
[illegible]	0	1	1	1	1	1

INTERNAL STATION CODE

	6	5	4	3	2	1
33	1	0	0	0	0	0
34	1	0	0	0	0	1
35	1	0	0	0	1	0
36	1	[illegible]	0	[illegible]	1	1
[illegible]	1	[illegible]	[illegible]	1	0	0
[illegible]	1	[illegible]	[illegible]	1	0	1
[illegible]	1	[illegible]	[illegible]	[illegible]	1	[illegible]
40	1	0	0	1	1	1
41	1	0	1	0	0	0
42	1	0	[illegible]	0	0	
[illegible]	1	0	1	0	[illegible]	0
44	1	0	1	0	1	1
45	1	0	1	1	0	0
46	1	0	1	1	0	1
47	1	0	1	1	1	0
48	1	0	1	1	1	1
49	1	1	0	0	0	0
50	1	1	0	0	0	1
51	1	1	0	0	1	0
52	1	1	0	0	[illegible]	
[illegible]	1	1	0	1	0	0
54	1	1	0	1	0	1
[illegible]	1	1	0	1	1	0
[illegible]	1	1	0	1	1	1
57	1	1	1	0	0	0
58	1	1	1	0	0	1
59	1	1	1	0	1	
60	1	1	1	0	1	1
61	1	1	1	1	0	0
62	1	1	1	1	0	1
63	1	1	1	1	1	0
64	1	1	1	1	1	

UHF CODE

EXTERNAL AND UHF STATION CODE

c. By making full use of the code table the number of stations the system can serve may be increased to:

1. 31 internal stations

2. 24 external stations

3. 8 UHF channels.

This fulfills the requirement that the system be capable of expansion if desired.

The second problem to be considered is whether the microphone should be supplied the direct current necessary for its operation locally or from a central source. When D-C is supplied from a central source, this is known as common battery operation. A common battery system is desirable because:

a. Simple circuitry, for provided the lines are not more than five or six miles long, it requires no amplifiers.

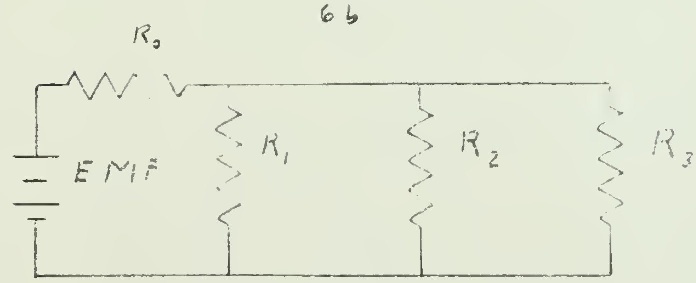
b. The current in the line can be utilized to send information to the central.

However, it has the disadvantage of requiring exacting electrical requirements on the line of which the most essential is that the source of D-C power have low, ideally zero, internal resistance. This is due to the fact that all circuits are in parallel to a common battery source as shown in figure 6a, where the resistances represent loads on the line. If the battery is ideal; i.e., no internal resistance, the current in any resistance is independent of the source. But if the source has a finite resistance, Ro, as shown in figure 6b, this is no longer true. Therefore, unless the common battery has negligible internal resistance, there will be everchanging current values in the individual speech circuits. This will result in noise and crosstalk in all speech circuits-- a highly un-

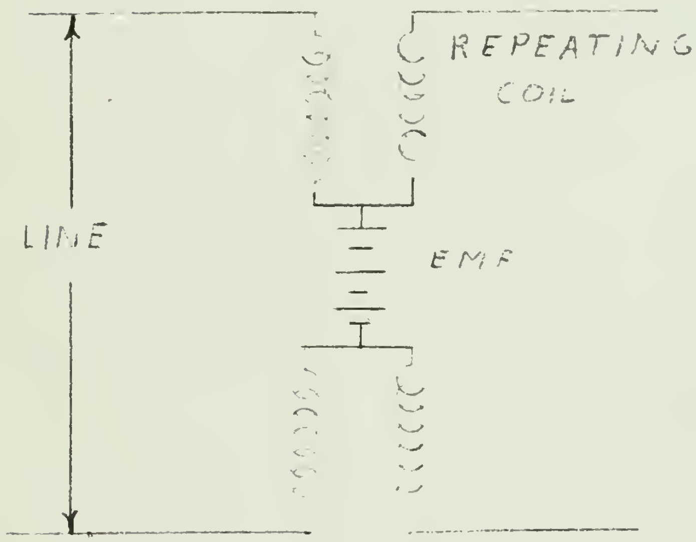
FIG 6

6a

6b



6C



Common Battery Connections

desirable result. The common battery may be given the appearance of having a negligible internal impedance by the use of repeating coils as shown in figure 6c, which also prevent crosstalk due to the fact that the common battery is common to all circuits.

The operator at each station should be able to control the audio level on the channel to which he is connected for several reasons:

a. to avoid loud noise when signal is strong

b. to enable weaker signals to be heard

c. to enable all signals to be heard at a comfortable audio level to avoid fatigue.

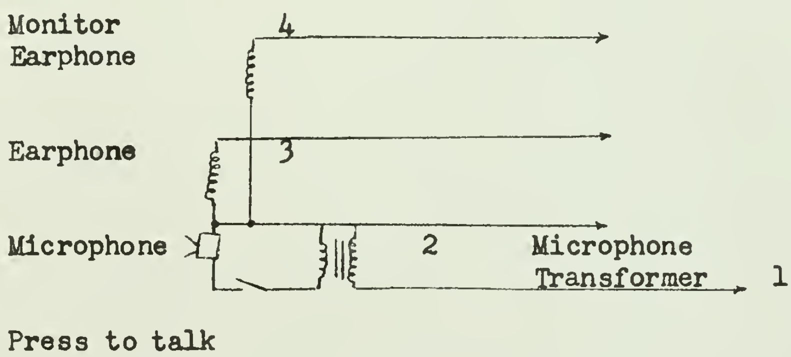
This control of audio level is difficult and wasteful of current to supply in a common battery system.

In view of these disadvantages the common battery system will not be used. A system employing audio-amplification which avoids these disadvantages and affords the operator better control over the audio level on his lines will be used. The audio amplification scheme has certain other advantages in allowing conference calls which will be discussed later.

The last problem to be decided is whether a single wire with ground return or balanced pairs should be used for the speech circuits. A balanced pair is a system that uses a metallic return path and is therefore balanced with respect to ground. The analysis of this situation as carried out in Appendix I indicates that the balanced pair is far superior to the single wire, hence it is the system to be used for the audio circuit.

A straight two-wire circuit is not too desirable, since to insert amplification in the line a hybrid or bridge transformer must be used as an amplifier is an unilateral device. This transformer must be a care-

fully balanced, well-designed unit, and to amplify both ways requires two amplifiers. In addition a hybrid transformer by its electrical nature inserts 3-6 db of loss into the channel. To avoid this a modified four-wire circuit is used as shown below:



Leads 1, 2, and 3 constitute the speech circuit, and leads 2 and 4 constitute the monitor circuit. Since it would be impossible to exactly balance leads 1 and 3 with respect to 2, there will be some crosstalk, but this is desirable in this case, since this provides a sidetone effect so that the operator can hear what he is saying. A press to talk switch is provided so that the microphone is dead unless the operator desires to use it. The microphone transformer prevents D-C from being on the line and matches the low impedance microphone to the line.

In summary then the additional system design features to be employed are:

a. A parallel D-C code.

b. The audio channels will employ audio amplifiers instead of a common battery circuit.

c. All audio channels will use a metallic return circuit instead of ground return.

Now that the system design philosphy has been established and the system parameters determined the system may be mechanized.

The mechanization of a system may be defined as the establishment of circuitry to accomplish the system design pholosophy established by considering the requirements the system must fulfill.

4. Mechanization of Station Panel.

The input device to the switching central is the means provided at the operator's location, referred to hereinafter as the station. This device must provide the following capabilities:

a. A means of selecting and calling any desired station in the system by a simple action such as pushing a button.

b. A means to signal the station that it is being called and identifying the calling station.

c. A means to enable the station to monitor any desired station's speech channel.

d. A means to select any desired UHF receiver transmitter.

e. A means to activate a transmitter (press to talk) if the speech connection is via an HF or UHF transmitter.

f. A visual indication of whom the station is in contact with at all times.

g. It is desirable to so arrange the input device that the operator or person using it may place any call he may desire without consulting an external agency such as a station directory or status board of some sort.

When all these capabilities are considered, it is seen that what is required is a panel of some sort with an array of buttons and lights to perform the desired functions. The input and output to the panel is the code trunk and the speech channels from the operator's headset. The press to talk function for the microphone is not on this panel, but is placed as a foot operated switch to enable the operator to have both hands free after placing the call.

An initial approach to this problem would be to represent every station that can be called in the system by a single lighted push button

which has two states of light control:

a. Blinking when that station is being called by the particular station indicated by the button label.

b. Steady when the station is in communication with the station indicated by the button.

In addition the button itself could be used to generate the missing stations code since the code is a parallel code utilizing

a. d-c present represents a binary "1"

b. no d-c present represents a binary "0".

To represent each station by a button would require a maximum of $62 \times 3 = 186$ buttons. The times three is due to the fact that station must be able to

a. establish a speech channel with another station

b. be able to monitor the station

c. be able to break the call down (an off function).

This multiplicity of buttons would be extremely confusing to an average operator and occupy a large amount of space. Thus it is clear that another solution should be sought.

An examination of the code table for this system, figure 5, reveals that in the first two digits the code repeats itself every four numbers. This characteristic of the binary code may be utilized to reduce the number of buttons required to call a station. The generation of the necessary code may be split into two parts with one set of four buttons utilized to generate the common elements of the code and another set of buttons to generate the remainder of the code. By using two sets of buttons it is possible to generate any desired code with a minimum of buttons. By arranging these buttons in a matrix arrangement and utilizing the signal lights to fill in the remainder of the matrix, a neat compact panel

may be achieved. In addition, the four buttons which generate the common elements of the code may be used to generate the talk-monitor bit of information.

This is the solution adopted and is shown in figures 7 and 8. This solution requires a maximum of

a. 16 buttons for 30 internal stations

b. 14 buttons for 24 external trunks

c. 8 buttons for UHF channels.

The UHF panel will be discussed later since it has additional features to be discussed in the section on UHF.

The sequence of operations for a station to call another station is:

1. The operator glances at the panel to see where the station is on the panel.

2. He presses the station button in the row associated with the station, then chooses talk or monitor mode in the column associated with the called station.

3. The called station's light comes on bright and steady indicating a call has been placed to that station.

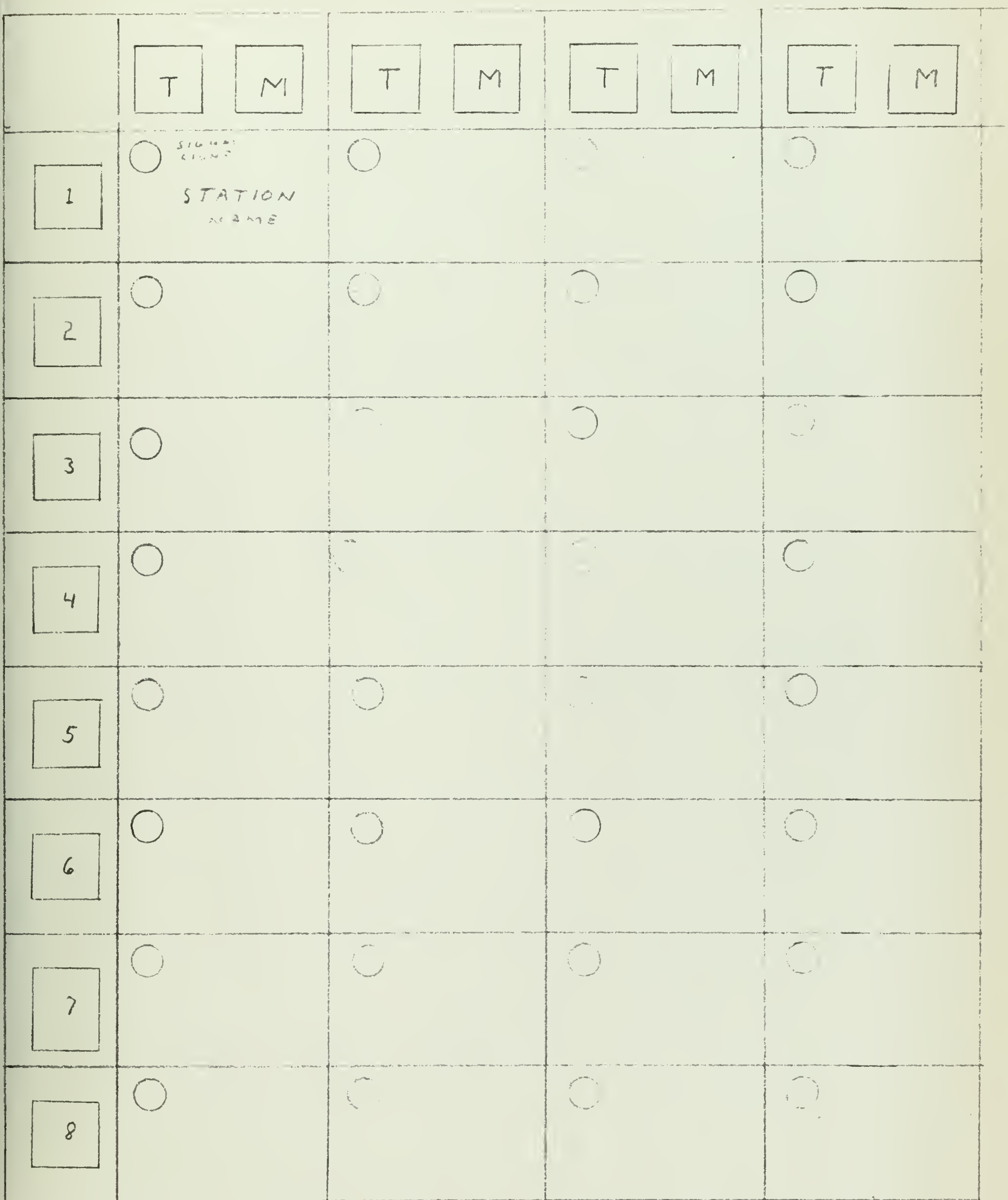
The wiring for the panel is shown in figure 9. The buttons are momentary make buttons which set the code in the register. The register is prevented from sending out the code to the code trunk until both buttons have been pushed generating the entire code. To generate the code the button contacts:

a. Put D-C on the code line for a binary "1" thus setting a 1 on the register.

b. Put nothing on the code line for a binary "0" thus setting a 0 in the register.

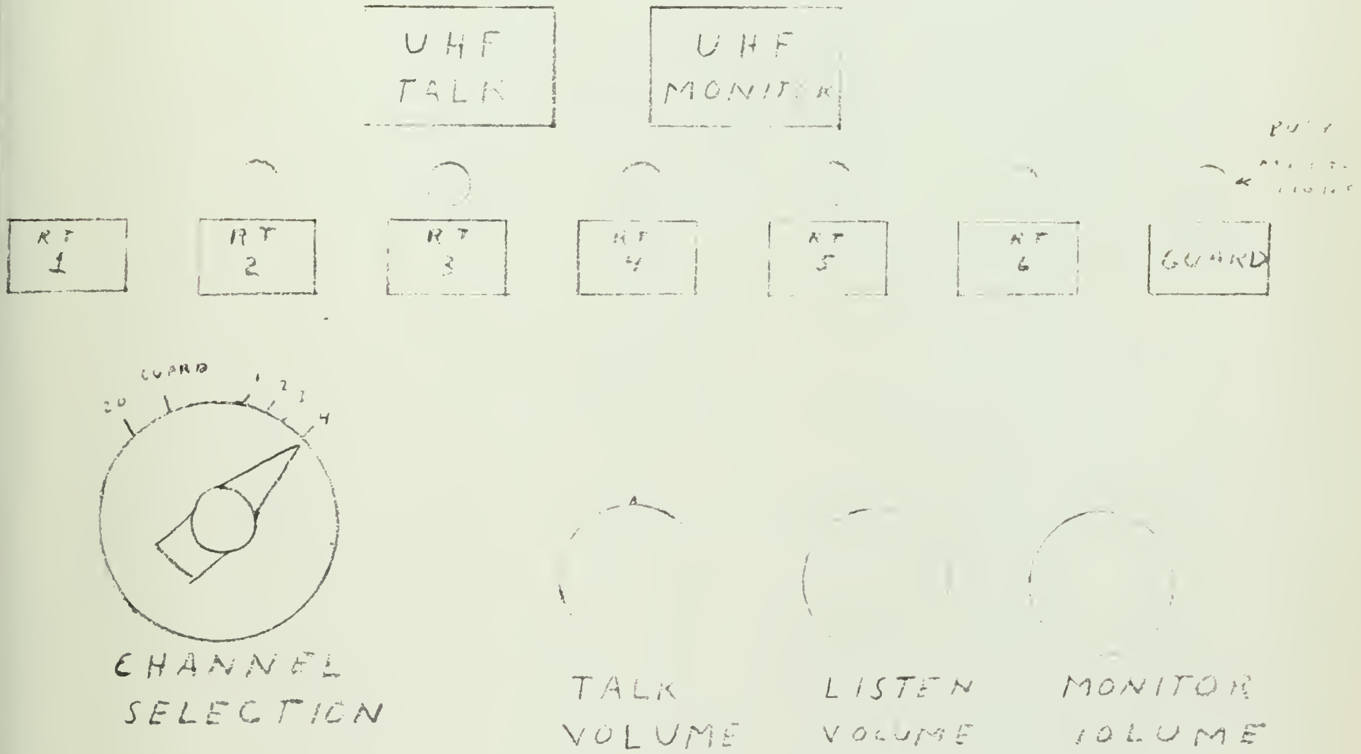
INTERNAL AND EXTERNAL
COMMUNICATIONS PANEL

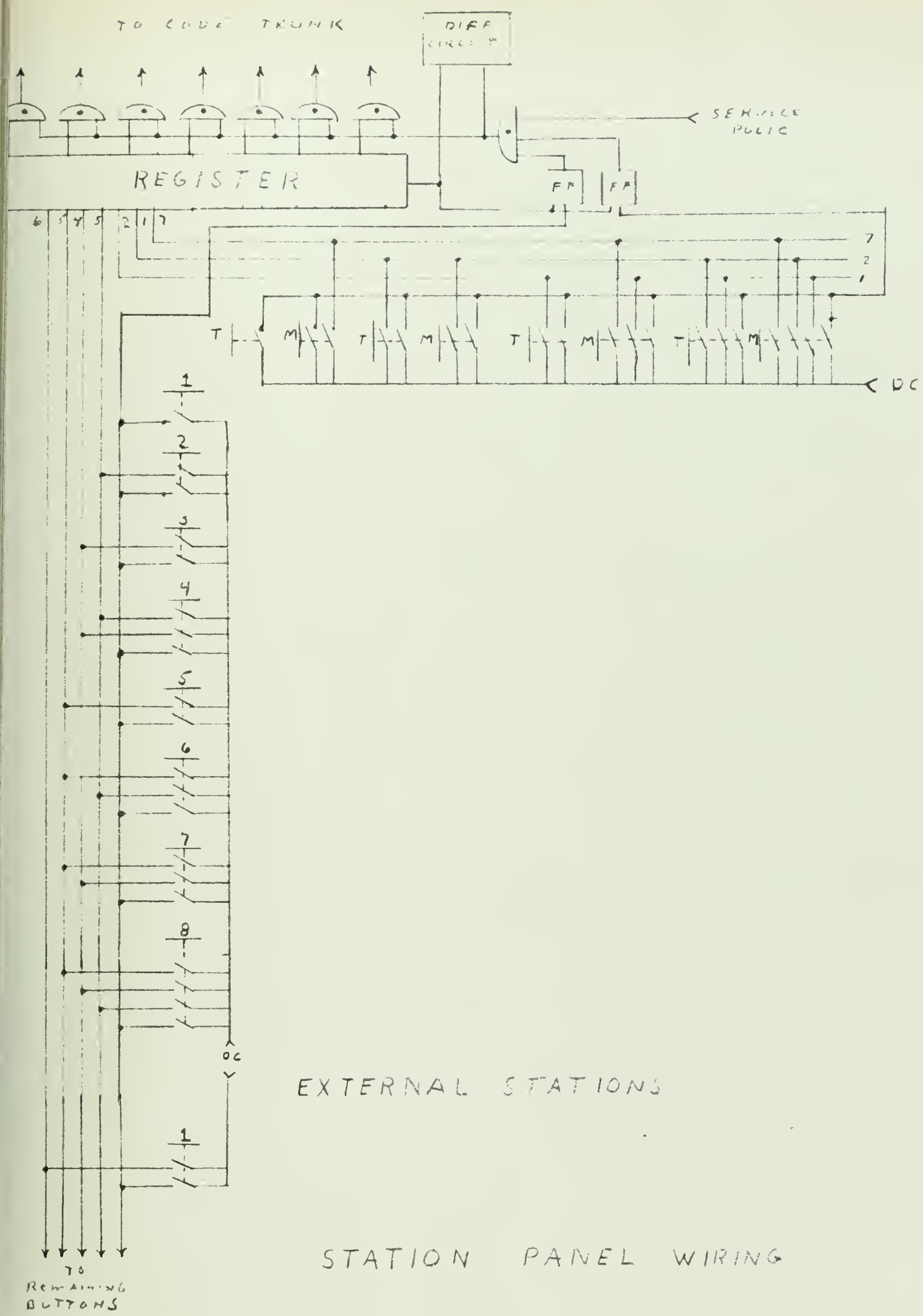
FIG 7



UHF PANEL

FIG 8

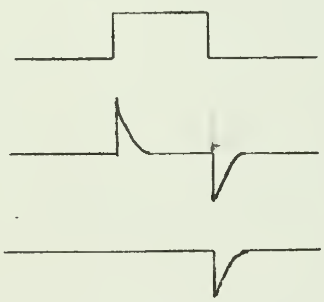




EXTERNAL STATIONS

STATION PANEL WIRING

The register is sent out to the code trunk only during the period of time allotted the particular station. This is done by generating a service pulse which opens the station's "and" gates during its allotted time interval. This service pulse may be derived from a counter which may be a part of the panel or, if several panels are used within an area such as helihut, a common counter may be used. This counter is a one through 62 counter and is slaved to a master counter which will be used in the switching central. Its design will be discussed later. The register must be cleared after every service pulse and this is accomplished by differentiating the service pulse as follows:



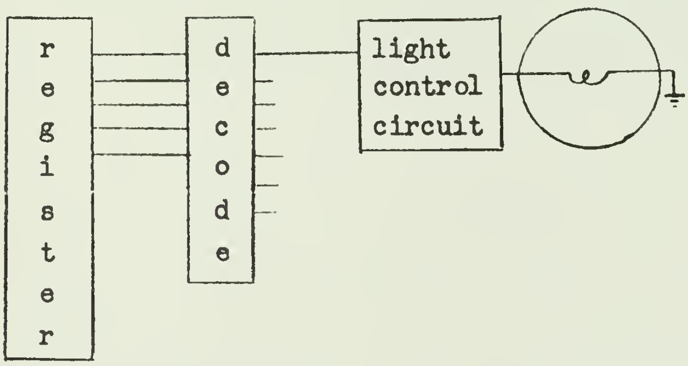
Service pulse

Differentiated pulse

Register clear pulse

The register clear pulse is then utilized to reset the register and service pulse flip-flops to zero.

The light associated with the called station is lighted by decoding the information contained in the register as follows:



and applying the decoded information to the selected light control circuit which actually turns the light on.

All the above enables the station to place a call, now the panel must become an output device and notify the operator when the station is being called. This may be done by making use of the fact that a common code trunk links all stations in parallel. Present on this code trunk, being sent to the central by the calling station, are the two pieces of information required at the called station:

a. the fact that the station is being called is signified by the presence on the code trunk of the called station code.

b. the calling stations identity is known by knowing what station's time slot is on the code trunks when the called station's code is detected.

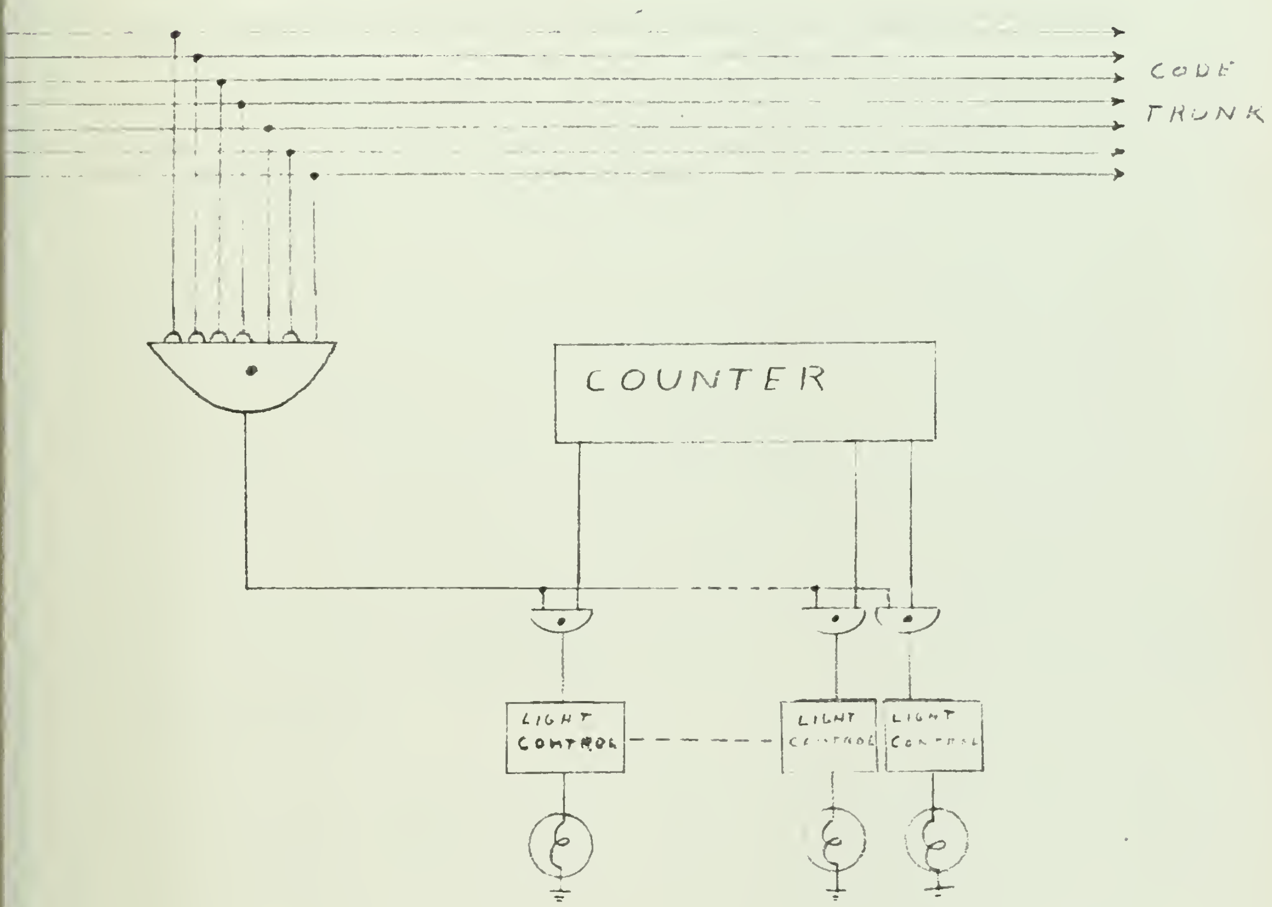
These two items of information may be used to light the desired one of 54 possible lights on the panel as shown in figure 10. To see the operation assume the called station's code is 000101. When this code appears on the code trunk the station "and" gate produces an output which is sent to an "and" gate in all the light control circuits on the panel. The counter identifies which station's time slot is present on the code trunk and the counter activates only one "and" gate of the 54 light control "and" gates at a time. At only one light on the entire panel are two signals present, one from the station "and" gate and one from the counter, which switches the light that identifies the calling station to the blinking mode. Therefore, the station is notified by this blinking light:

a. that it is being called

b. by what other station it is being called.

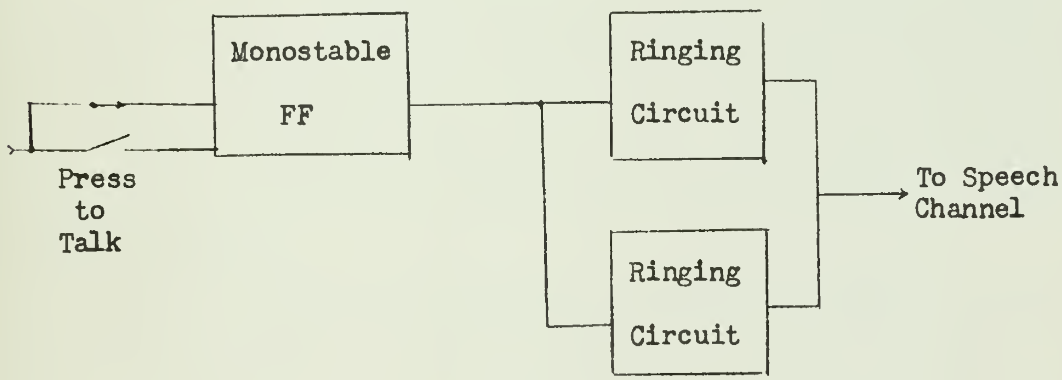
Another function that must be supplied by the station panel is a means of activating a transmitter when the channel connection desired is via an HF or UHF transmitter. Analysis of this problem reveals that the

FIG 10



INCOMING CALL SIGNALLING

function required is a means to turn on the "press to talk" circuit of the transmitter when the transmit function is desired. It would be impractical to run a wire from each transmitter to each station to accomplish this function. A way to accomplish this function would be to use a pulsed signal to activate the press to talk function. It would be highly desirable to send the pulse information via the speech channel to avoid the use of extra wiring. Considering all this leads to the conclusion that voice frequency pulses or tones may be used to turn the transmitter on and off. However, when this is done care must be taken to avoid having the transmitter turned off by speech frequencies which resemble the tones used to turn the transmitter on and off. This method of signaling has been used successfully before[9] and the problems associated with speech imitation investigated. By the use of two different frequencies and careful selection of the frequencies to be used, this method of turning the transmitter on and off may be successfully utilized. The circuit in block diagram form is shown below.



The operation is as follows:

a. When the press to talk switch is pressed the monostable FF changes state causing the tone generator circuits to ring, generating the required tones.

b. These tones are put on the station's speech channel and the monostable FF flips back to its next state, allowing the ringing circuits to restore.

c. At the transmitter the tones are detected and used to close the relay turning the transmitter on.

d. When the press to talk switch is released the above action is repeated turning the transmitter off. A simple circuit can be constructed and placed at each transmitter connected to the system to enable this operation of turn on and turn off.

The last function to be provided is a ringing button. Ringing is not done in this system, for the lights provide all the indication that is needed, plus the fact it would not be desirable to ring on a station's line if that station is already talking to another station when it is called. Therefore, ringing is provided as an optional function only. Since the system employs amplifiers, ringing must be voice frequency ringing and 1000 cycles was chosen as the most desirable tone to use. This tone is chosen because it is a pleasing tone and it can be heard more readily than other tones[10]. This ringing tone is generated by using a ringing circuit which is caused to ring by pushing the ring button, thus producing only one tone for each push of the button. This prevents prolonged ringing and simplifies circuitry.

The last circuitry to be provided on the station panel are the volume controls for the speech and monitor channel. As will be seen later, the amplifiers are located at the central. Thus a variable pad is provided at the station to enable the operator to control his volume levels.

It is to be noted that no method has been provided specifically to break down an established call. To accomplish this would require one more

bit of information to be sent to the central; hence, another code wire. Also, another button would have to be provided and additional circuitry. To prevent all this, advantage is taken of the intelligence of the operator, and a call is broken down just like it is established, by pressing the same two buttons. As explained in Section 5, the central will detect the second establishment of the call and break the connection.

5. Mechanization of Switching Central.

As was pointed out earlier all channels are terminated in the switching central and the channel connections may be arranged as an array of crosspoints, where a crosspoint is defined as a point of connection of two channels. At all points in the array where a connection is desired, a switching device of some sort must be used to make the connection. The properties of this device must now be determined for the choice of device will influence the logical circuitry necessary to make the connection upon demand.

Very rigid requirements must be met by this switching device with respect to crosstalk, noise and speech transmission, such as:

a. the impedance ratio between open and closed position must be very high in the audio range used in the speech circuits.

b. it must be able to handle reasonable amounts of speech power.

c. it must be highly reliable.

d. it must be inexpensive for in this system a large number of them will be used.

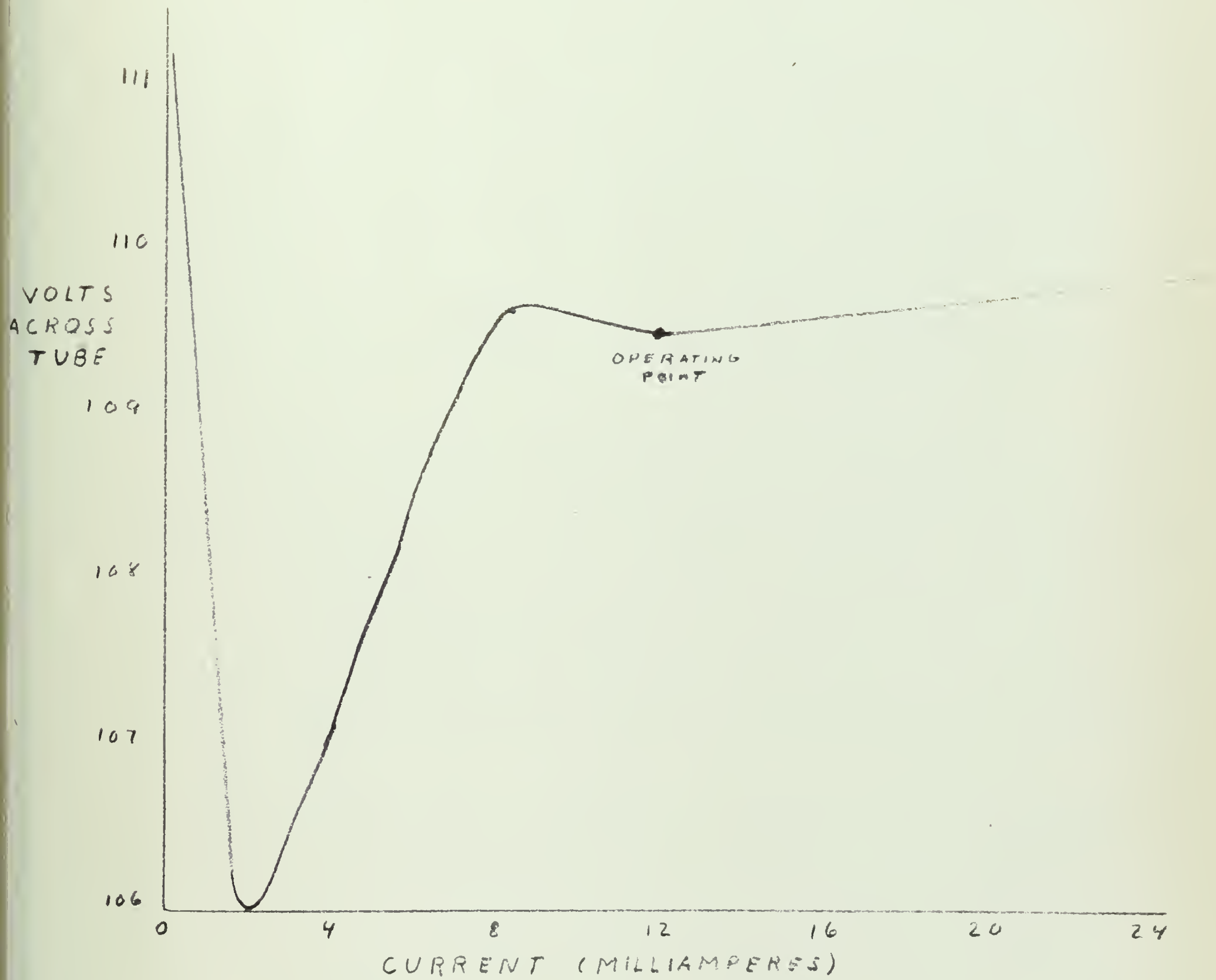
A satisfactory solution exists and has been used for a long time in the form of relay operated metallic contacts. Closed, the contacts are equivalent to a .01 ohm resistor; open, they appear as a capacitor of a few micromicrofarads, thus presenting an impedance of approximately 100 megohms at 1000 cycles. Numerous contacts may be provided to accomplish other functions besides completing the speech circuit. Also they have been used for so long that they are a highly reliable item at a fairly low price. However, they have the disadvantage of slow operation, approximately 10 milliseconds, and requiring fairly large amounts of power to operate. In view of these disadvantages it seems appropriate to seek other means of switching to see if there is a better method.

At the present time a large variety of electronic switches is known, such as: vacuum tubes, gas tubes, saturable inductances, semiconductor diodes, and transistors. Vacuum tubes may be eliminated from consideration as the power required to heat their filiments would be excessive. So far it has not been possible to build with magnetic materials devices having a sufficiently high impedance ratio between the conducting and non-conducting states. Thus, saturable induction may be eliminated from consideration. The most promising devices to consider are cold cathode gas tubes and semiconductor diodes and transistors.

A cold cathode gas tube has been developed by Bell Telephone Laboratories for use in electronic switching[11], and means have been developed for its use in an automatic switching central[12]. The static characteristic for such a tube is shown in figure 11. The breakdown voltage is 190 volts and the sustain voltage for a 12 milliampere current is 109.5 volts. At this operating point the dynamic resistance of the tube is a -180 ohms at 1000 cycles, which is a decided advantage. It can handle powers of 12 milliwatts into a load of 2000 ohms. The firing time of the tube is approximately 100 microseconds, and is stable due to the introduction of radioactive bromide to assist the ignition breakdown of the gas. The tube generates a noise no greater than 10^{-11} watts in the audio band. Reference 13 describes other gas tubes developed for switching, but they are in general not as suitable as the Bell Laboratories' tube. However, the negative resistance of the tube at its operating point is dependent on the ambient temperature of the tube, thus a fairly high load resistance is required to absorb these variations in resistance. This high load resistance aggravates the crosstalk problem.

Semiconductor diodes can be used as contact points in telephony, and

FIG 11



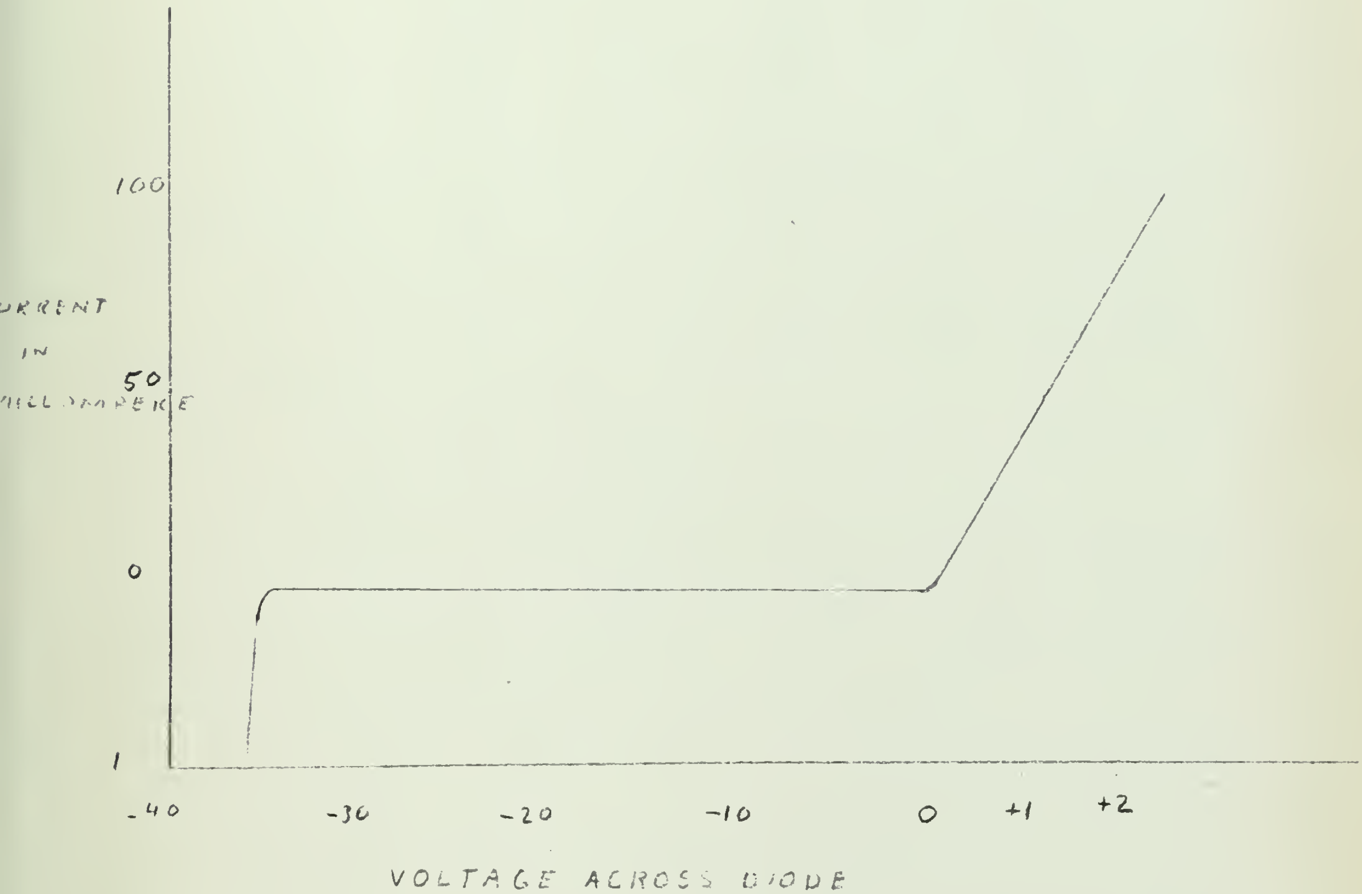
STATIC CHARACTERISTIC OF GAS DIODE

have good possibilities. Figure 12 shows a typical static characteristic of a P-N junction diode. In the conducting direction with one volt applied the current is approximately 50 milliamperes, and the slope corresponds to an internal resistance of approximately four ohms. In the nonconducting region the resistance is about one hundred megohms until over thirty volts is applied, at which time avalanche breakdown occurs. The parasitic capacity is about three to four micromicrofarads. It can easily handle a power of fifty milliwatts into a 600 ohm load. A fully automatic telephone exchange has been constructed utilizing P-N junction diodes[14].

Symmetrical transistors may be used as a switching device[15]. They have been used as a switching element in an automatic switching system supplied to the U. S. Army[16] and U. S. Navy[13], Insufficient information could be obtained concerning their use as a switching element in such applications to fully evaluate their suitability for use in this system. Therefore symmetrical transistors were not considered for use as a switching element.

In the devices mentioned above only the gas diode possesses a memory function. The P-N junction lack a memory function so they must be complemented with devices that will perform a memory function before they are usable. The semiconductor devices have a switching time of as low as one microsecond compared with 10 to 500 microseconds for gas tubes and one to ten milliseconds for a relay. These characteristics may be summarized in the following table:

FIG 12



PN JUNCTION DIODE CHARACTERISTIC

	Impedance Ratio ohms	Speed of Operation Microseconds	Power Handling Milliwatt	Power Required to operate Milliwatt	Life and Reliability	Memory Function	Number of paths closed by switch
Relay	10^{10}	10,000	1000	300	High	Yes	Many
Gas Diode	10^{10}	10-500	100	Very Low	High	Yes	1
P-N Junction Diode	10^{11}	1-10	10	10	High	No	1

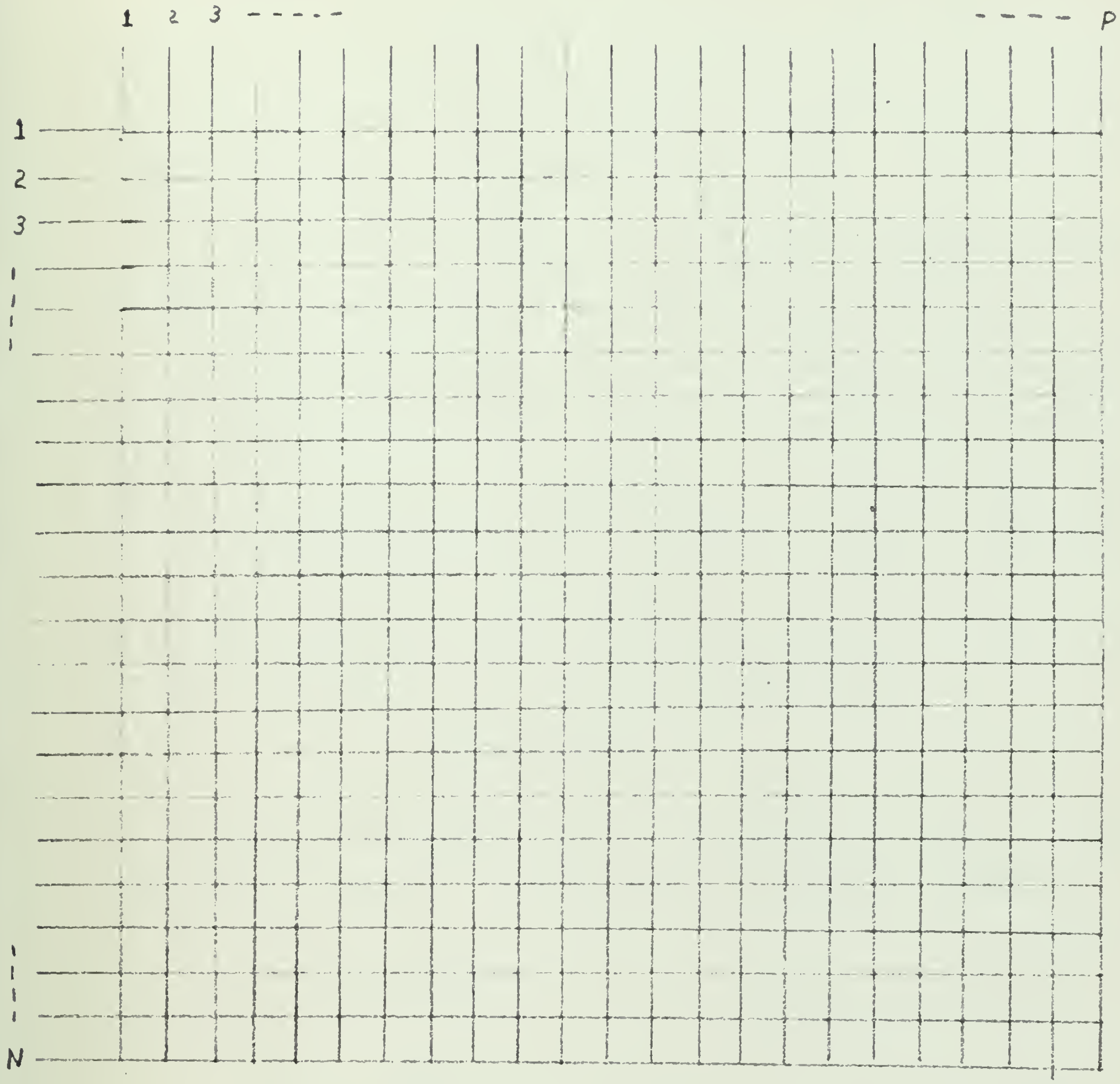
Consideration of this table and the characteristics listed above narrow the choice down to the relay and the gas diode for their qualities of power handling, life, and memory function. The choice between these two is extremely difficult, but the relay is choosen because:

a. It is desired to switch several paths at the same time, which cannot be done without using a gas tube for each path.

b. The relay is in general a more economical device than the gas tube.

Now that the crosspoint element has been selected, the switching central may be designed. It is convenient to use crossbar representation in discussing switching diagrams regardless of the system being considered. A crossbar diagram as shown in figure 13 consists of an array of N horizontal wires and P vertical wires where each wire or line represents a communication channel. In this system $N = P = 60$ wires, so it is a square array. Each of the crosspoints in the array controls a gate which makes or breaks the connection controlled by the crosspoint. In this system two such arrays must be considered: one for the speech channel switching or talk

FIG 13



CROSSBAR DIAGRAM

matrix, and one for the monitor switching or monitor matrix.

The number of actual speech connections or relays required in this N x P matrix may be determined by considering the matrix in view of what speech connections are desired. When the array of crosspoints is considered as a matrix it is seen that

1. points on the diagonal may be disregarded since these represent a connection of the channel with itself which is not required.

2. the matrix is symmetrical about the diagonal. That is, an x,y point is identical to a y,x point as far as the speech channel is concerned. It is immaterial if x is connected to y or y is connected to x.

For a symmetrical matrix $N = P$ this may be expressed as $\frac{N \times P - N}{2}$.

Thus a speech connection need be provided for only this number of points for full flexibility. If $N = P = 60$, this will require:

$$\frac{60 \times 60 - 60}{2} = 1770$$

That this is the total number of connections required for 60 channels taken two at a time may be checked by the conbinational formula:

$${}^{n}C_{r} = \frac{n!}{r!\,(n-r)!}$$

$${}^{60}C_{2} = \frac{60!}{2!\,(60-2)!} = 1770.$$

This would be the total number of speech connections required for full flexibility in the talk matrix.

Before a particular crosspoint design may be considered the problem of selection of the crosspoint must be considered. Since the crosspoints are arranged in an array, the problem of selection of a particular crosspoint upon demand is very much like the problem of core selection in a core memory scheme. They both demand fast, random access to any point in

the array. To choose a particular connection in the array of crosspoints two things must be known:

1. Identity of calling station,

2. Identity of called station.

The identity of the calling station may be obtained from a counter in the central which keeps the time slots in their correct order. This counter, which counts up by one each time period, may be used to activate the N horizontal wires in the array one by one up to N, at which time it starts the cycle over again. Thus only one of the N horizontal wires has a voltage E_n on it at a particular time, which identifies the calling party. The P vertical wires may be activated by decoding the binary code filed in the time slot, which is on the code trunk at a particular time. Thus, according to the code present, only one of the P vertical wires has a voltage E_p present on it at a time identifying the called party. Therefore, at only one crosspoint in the entire array is there a voltage $E_n + E_p$ present which identifies the crosspoint to be closed. Thus, if A calls B the sequence is:

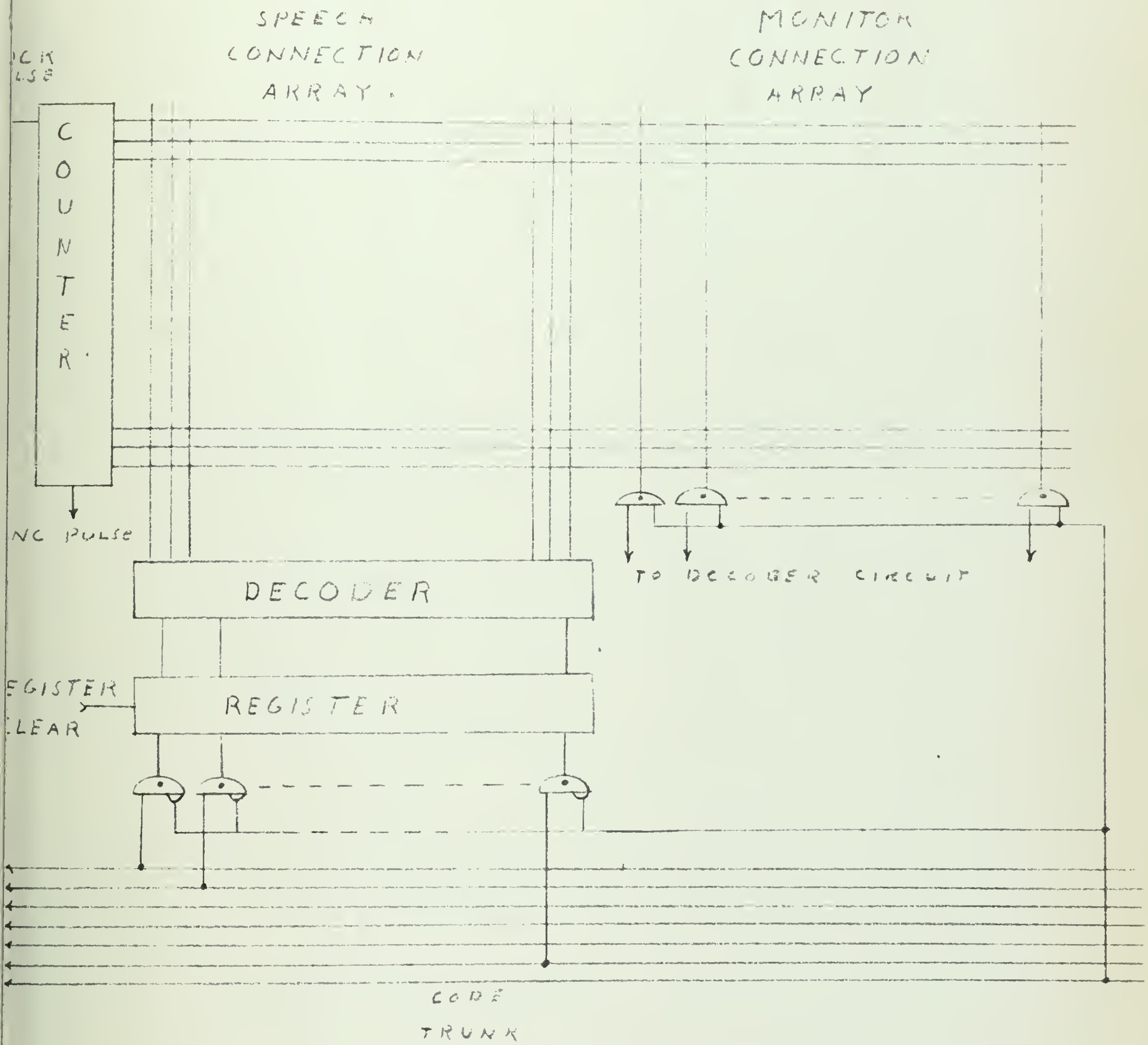
a. when A's time slot occurs the wire corresponding to A in the horizontal array has a voltage E_n placed on it by the counter;

b. the binary code present in A's time slot corresponding to B is fed into the register where it is decoded, and B's wire in the P vertical array of wires has a voltage E_p placed on it;

c. only at the A-B crosspoint is a voltage $E_n + E_p$ present, thus selecting that crosspoint as the gate to be closed. This is shown in figure 14.

Since a counter as used here in the selection circuitry has a tendency to skip counts, which would throw the whole selection process out of step,

FIG 14



TALK MONITOR CODE LINE - LINE 7

0- TALK

1- MONITOR

CROSSPOINT SELECTION CIRCUIT

it is timed by an accurate clock pulse generator to insure that this does not happen. The switching central counter is also used to step the local counters to insure that this does not happen at the local station counters also. No local counter, therefore, steps to the next count unless a pulse arrives from the master counter at the switching central. Further, when the master counter reaches the last count and starts the cycle over again, a double pulse is sent out to all station counters to insure that all counters start the count cycle at the same time. The cycle time of the counter is set at 0.1 seconds. This figure was arrived at by compromising between setting the cycle time fast enough so the station cannot place calls faster than the system can accept them and not making the cycle time so fast that special techniques must be used to handle the service pulses. In addition, too fast a cycle time would increase the complexity of the counters, which is not desired. If the operator was able to place two calls before the service pulse arrived, it is obvious a wrong number would result. For example, if he called

B, whose code is 0000101

C, whose code is 0000110

the call the central would get is 0000111, which is neither B nor C. In view of all these factors a 0.1 second cycle time was chosen as a good compromise.

The same selection circuitry may be used for both the talk matrix and the monitor matrix, since only one or the other is used in any one time interval. The selection pulses are gated from one matrix to the other by the state of the talk-monitor wire in the code trunk as shown in figure 14.

The requirements as stated in part 2 and the convention adopted in part 4 of this paper regarding how to break down or disconnect a placed

call impose four states of logic on the relay used to make the speech connection for internal stations. The four states of logic are, assuming A calls B:

a. A calls B, selection of A-B crosspoint—do not make speech connection;

b. B answers A's call, selection of B-A crosspoint—makes speech connection;

c. A breaks connection to B by repeating initial procedures used to call B. Release (break) speech connection;

d. B breaks connection to A by same action as in c.

In effect step d is a redundant process, as what would be desired is that only one of the two parties involved in the call be required to release or break the call.

A consideration of these four states of logic reveals that what is required for the first two steps is an "and" function, and for the last two steps an "either or both" function. Since the crosspoints involved have a voltage $E_n + E_p$ applied only for the duration of the time slot or $\frac{.1}{62} = 1.612$ milliseconds, a memory function must be provided to remember that one of the crosspoints has been selected so that when the second crosspoint is selected the connection will be made. A simple, reliable, economical memory device is a cold cathode gas diode such as those used in many telephone switching circuits[17]. These tubes require a fairly high plate voltage for their operation, but advantage of the margin between the firing voltage and the extinguishing voltage the tube may be taken. By biasing the tube at a point between these potentials, a much lower pulse of voltage may be used to turn the tube on. Also, if this pulse, which is a positive pulse, is switched to the cathode after the tube is fired, the

same pulse may be used to turn the tube off.

Consideration of all these factors results in the design of the logic for making and breaking the speech connection as shown in figure 15. The sequence of operation is, assuming A calls B:

1. The A-B crosspoint is selected generating a positive pulse which fires the associated gas tube;

2. When B answers, the B-A crosspoint is selected firing the associated gas tube;

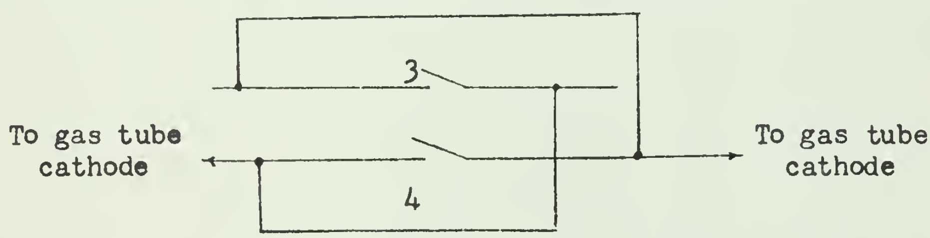
3. Both gas tubes are now on turning the "and" circuit on which activates the relay;

4. When the relay closes the crosspoints are switched to the cathode of the gas tube;

5. When A calls B again selecting the A-B crosspoint the tube is turned off, opening the relay;

6. When B breaks down, the second leg of the gate is turned off.

By cross-wiring relay contacts 3 and 4 as follows:



either station may turn both tubes off, but then a convention must be adopted as to which station breaks down the call, for if A should break the connection and then B should break, the break by B is treated as if a new call is being placed. This convention is, that the station originating the call breaks the call down.

The monitor array requires a relay at each crosspoint for full monitoring flexibility; that is, for an internal station to be able to monitor every other channel in the system. Since this would require a maximum of

30 x 62 = 1860 relays, it is obvious that full monitoring capability must be limited to a few stations, and the other stations be limited in what channels they may monitor. In this array only two states of logic are required, for the relay must be closed upon the first selection of the crosspoint. The logic for this situation is shown in figure 15b and is a simplified version of the logic used in figure 15a. This same logic is used to close the connection to the UHF receiver-transmitter in the talk array. It is to be noted in both schemes of logic that the gas tubes serve as a buffer between the relative fast pulse of the crosspoint selection and the slow "pull-in" time of the relay. Thus a relay with slow "pull-in" times on the order of 10 milliseconds may be used as the time of relay closure is insulated from the cycle time of system.

The fact that the speech connection relay circuits are independent, requiring only a wire connection of the crosspoint selection circuitry, may be used to some advantage. The crosspoint selection circuitry is easily built and in fact is built to select any of the N x P - N crosspoints in the matrix. However, it may be that not every speech connection will be desired or the full number of stations may not be present in some cases. The speech connection circuits may then be built as plug-in units and supplied only to the extent needed. In addition this simplifies maintenance as the speech connection circuits may be easily removed and tested.

When an external station is to be called a different technique must be used. These stations cannot be on the code trunk directly since, in general, they are at a remote location and are reached via a radio line such as radio relay, scatter, or H.F. radio. Therefore, the D-C code must be converted into a form whereby the code may be sent to the external station to notify

FIG 15A

SPEECH CONNECTION LOGIC

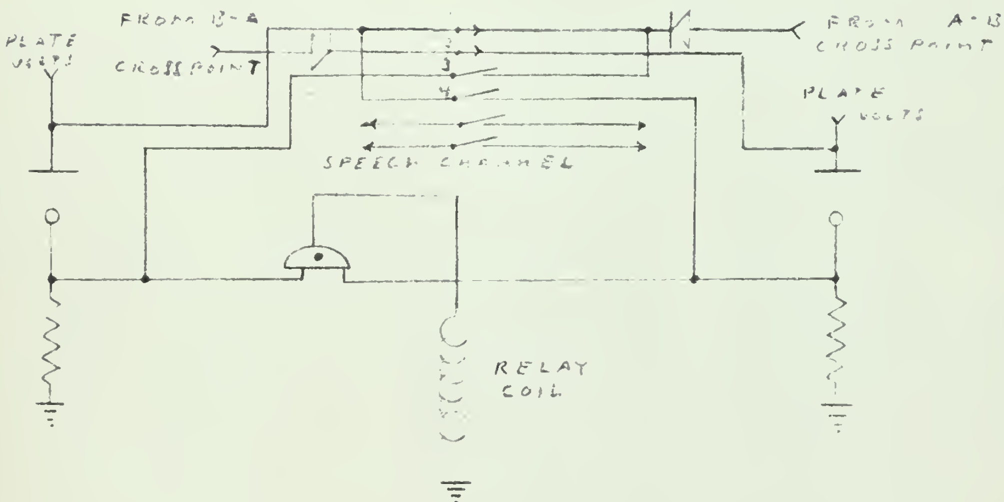
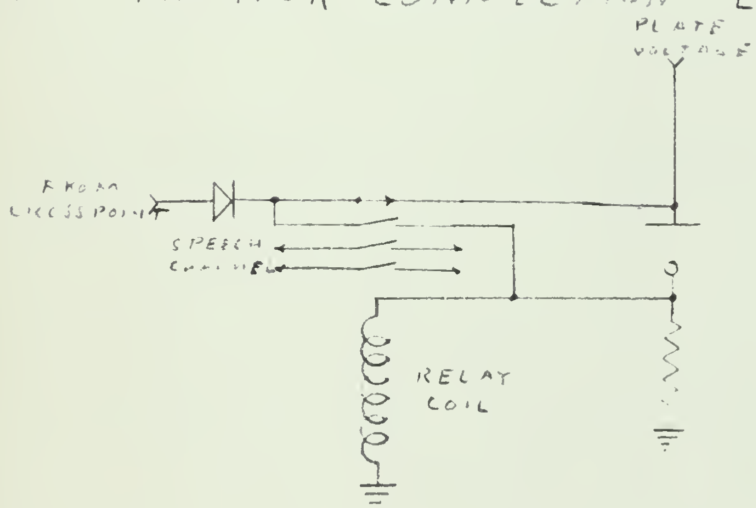


FIG 15b MONITOR CONNECTION LOGIC



a. it is being called

b. who is calling.

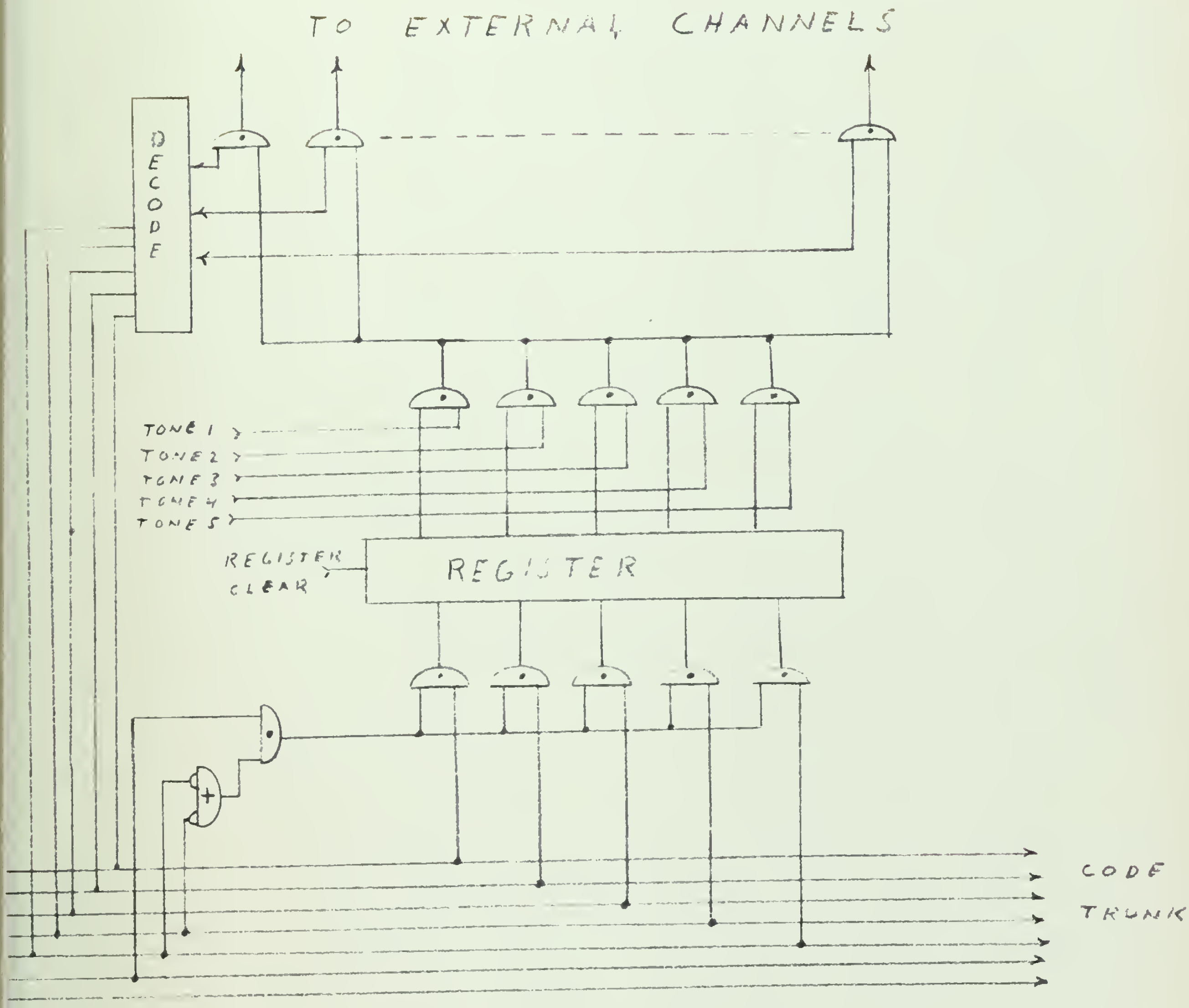
A connection to the external station exists in the form of the communication channel to that station for communication cannot be done unless this link exists. So the obvious solution is to use this speech channel itself to send the signalling information. To do this the D-C code must be converted to frequency pulses or tones and the frequencies so selected that they do not interfere with each other. A method for choosing these frequencies is discussed in Appendix II. Now that the signalling scheme has been selected it must be mechanized. The solution to this problem is shown in figure 16 and is discussed below.

From the code table, figure 5, the logical statement for determining if station called is an external station is $6\,\bar{5}\,\bar{4} + 6\,5\,\bar{4}$. In truth table form this is:

6	5	4	Out put
1	0	0	1
1	0	1	1
1	1	0	1
1	1	1	0

which may be reduced to $6\ (\bar{5}\ \bar{4})$ since an output is desired if line 6 has a "1" and if either lines 5 or 4 is a "0". This determines the configuration of the logic circuitry to enable the code on the code trunk to be read into the register. A register is used so that a longer tone than 1.6 milliseconds may be sent. The register is cleared by the double pulse generated when the counter starts a new cycle. This allows a 100 millisecond tone to be sent, which assists the detection circuitry in avoiding being triggered by speech sounds. The register then opens the appropriate tone "and" gates allowing a tone for each binary "1" in the register to be

FIG 16



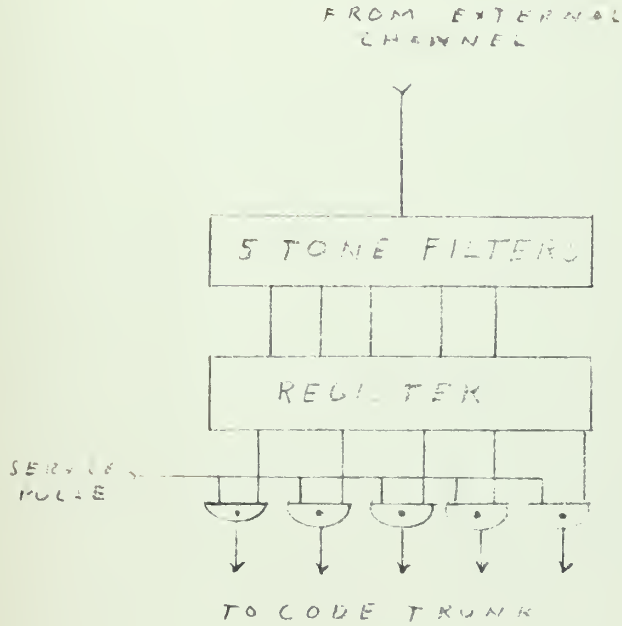
EXTERNAL CHANNEL SELECTION LOGIC

sent. Since the tones are frequency multiplexed they may be sent on a common wire to the gates for each external channel. Since each external channel is assigned to a specific station the code also tells which channel is to be used. Therefore, decoding the code selects the gate to be opened to allow the tones to be placed on the desired external channel.

The external channel may be treated exactly as if it were a station on the internal system, for in effect it is. When an external station calls a station on the system it closes the speech connection relay on the first selection of the crosspoint in the talk array, thus extending this station into the other system. Therefore, the channel is serviced to determine if it has a call present for a station on the system. This is accomplished by providing a set of five filters and a register for every channel and servicing the register for calls as shown in figure 17. Although this requires redundant circuitry, the circuits are simple, limited in number, and it is better to provide several simple circuits than one complex circuit, such as a line scanner. Circuitry is not provided to send the tones via HF radio, for here it is assumed that when an HF radio set is used it is continually monitored by the stations involved.

It is to be noted here that the external station is denoted by the sixth bit being a binary "1". The external station may itself be a station on an automatic switching central at its location. Thus on one system it is an internal station, on the other system it is an external station. The sixth bit keeps this situation in order. What the counter really tells the called station is that a call exists for that station on a certain channel which goes to a remote location. Now if only one station is assigned at each end of the channel the light that denotes that channel may be labeled with the station name and it tells the station it is being

FIG 17



ONE IS PROVIDED FOR EACH EXTERNAL CHANNEL

External Channel Logic

called by the station at the other end of the channel. However, the system flexibility is such that more than one station may use this channel to the external location and at the other end more than one station may receive calls via this channel. In this situation the individual stations may be called selectively but they cannot be told which one of the several stations at the other end of the channel is calling since all the information the counter provides is what channel the called station must connect to, to receive the call. In addition, when several stations may use an external channel they must monitor the channel first to determine if the channel is busy before placing a call via the channel. This is necessary as the busy-idle state of the channel is not provided as an output to any station. This is not considered as a serious omission since all operators who may use the channel have the capability to monitor the channel when desired.

In the switching matrix the audio channels are not brought in symmetrically. The 30 internal channels are brought in both horizontally and vertically, but the external and UHF channels are brought in only vertically. This is done because there is no reason for the external and UHF channels to do any switching in the matrix. Only the internal stations are switched <u>to</u> the external and UHF lines.

The external and UHF channels are not switched to the internal stations. This is an important difference. With this distinction in mind the number of relays required for speech connection for a full system of 30 internal, 24 external and 8 UHF stations may be computed.

a. Internal relays required $\frac{30 \times 30 - 30}{2} = 435$

b. External relays required $30 \times 24 = 720$

c. UHF relays required $30 \times 8 = \underline{320}$

1475

1475 total relays are required for full operation of the system. Of course, this number of relays may be drastically reduced by limiting the flexibility of the stations. A typical example of use may be as follows:

a. assume 20 internal stations

b. assume 20 external channels distributed:

1. 2 stations may call 12 external stations each
2. 10 stations may call 4 external stations each
3. 8 stations may call 2 external stations each.

c. only 15 stations may use UHF

Now the total number of relays required becomes:

a. Internal relays required $\frac{20 \times 20 - 20}{2} = 190$

b. External relays required $24 + 40 + 60 = 80$

1. $2 \times 12 = 24$
2. $10 \times 4 = 40$
3. $8 \times 2 = 16$

c. UHF relays required $= 15 \times 8 = 120$

390

390 total relays are required by this situation.

Therefore, the number of relays used in the speech relay is dependent upon the flexibility demanded and is easily supplied by merely adding or taking away relays, in the speech or talk matrix.

The monitor matrix requires a relay for every channel the station must be able to monitor. Here also, by the use of plug-in relay circuits the flexibility of system may be utilized. If a demand for monitoring a channel arises, a relay circuit is plugged in and the station may monitor. So the monitor capability may be supplied or denied as desired.

As stated in the requirements there is a need for an output indicating when a speech connection is made. This is for use when the system is used

with data processing equipment utilizing a computer. It may be that when two stations are in communication with each other they desire to pass information from this data processor from one station to the other. To do this the data processor or computer must know two things:

a. is passage of information desired

b. which station is talking to which station.

This process may be accomplished as shown in figure 18. To provide this output which may be called a communications mark, a pulse is generated for each of the stations concerned, but no two pulses occur at the same time. Only 15 stations, hence 15 pulses, are shown in figure 18 for convenience. A contact on the speech connection relay is utilized to perform the desired function. The operation is as follows:

1. station talking desires to send data so the station closes the communication mark switch;

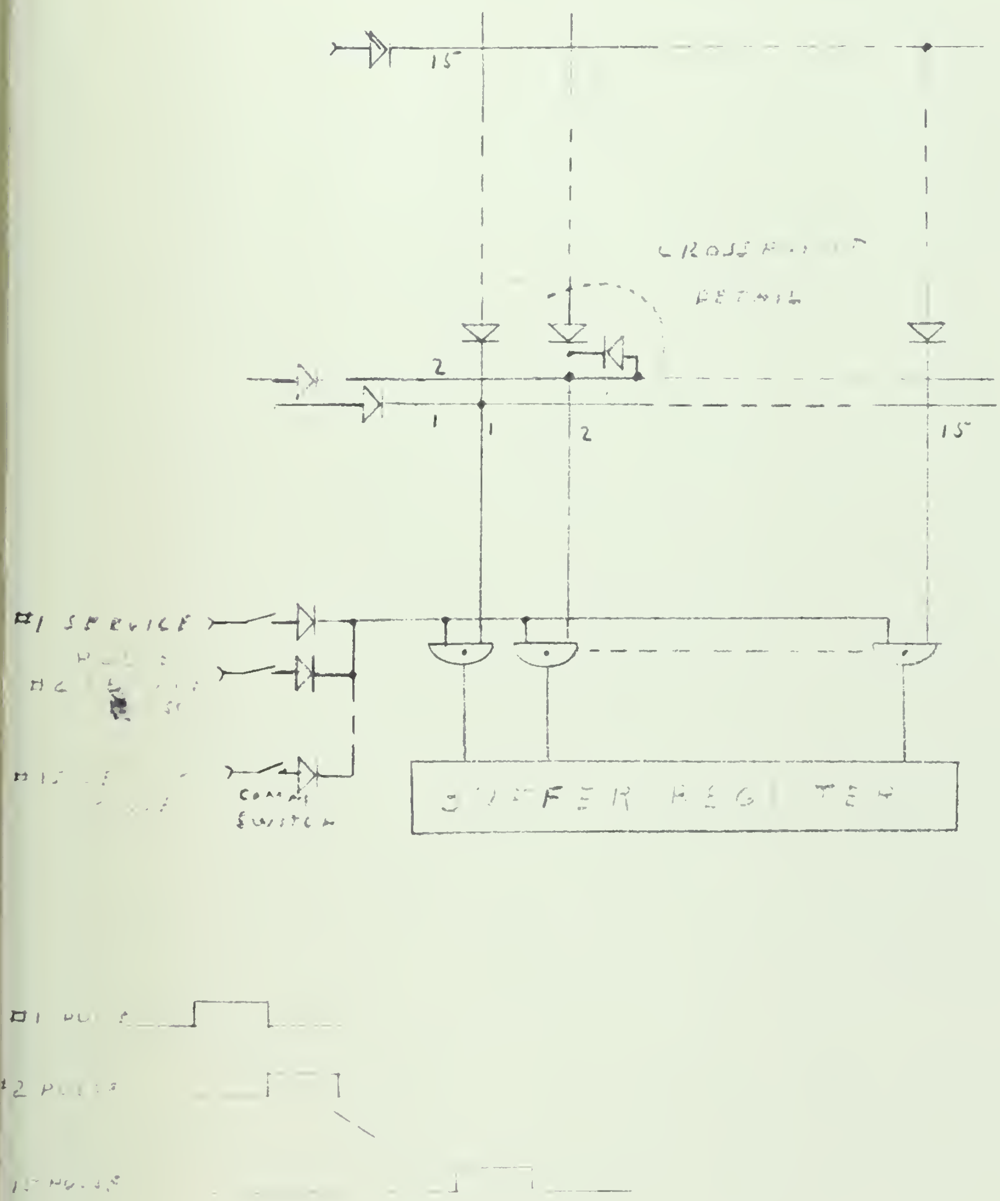
2. when that station's communication mark service pulse occurs it is sent up through the matrix and comes back out on as many wires as the station is in communication with;

3. both legs of the "and" gate are on, so a "1" is filed in the flip flop corresponding to that wire the pulse came back out on;

4. this information may then be utilized to tell the computer to send data to these stations.

The diodes shown in the detail in the matrix are isolating diodes to keep the positive service pulse from feeding through closed connections and giving false information. The communication mark service pulse is supplied by the associated computer.

FIG 18



COMMUNICATION MARK MATRIX

6. Audio Channel Design.

The audio channels brought into the central are a three-wire system with a monitor channel superimposed upon the three-wire system as described earlier. At the central the common return wire is grounded, forming what is called a station ground. This avoids the necessity for switching the common return at the relays. It does increase the susceptibility of the system to crosstalk, but if care is used in avoiding long runs and too close a coupling of wires in the central, this should not be a serious problem.

The connection to be made is as shown in figure 19. This is accomplished by running the audio channels as shown in figure 19, thus the correct connection is always made. The system of operation chosen requires audio amplification on each speech channel. To provide the necessary amplification a technique of analog computation may be used that of the summation amplifier. This amplifier has several desirable properties which make it desirable for this use, such as:

a. gain is relatively insensitive to variations in the amplifier

b. a virtual ground is established at the input to the amplifier, enabling several channels to be connected to the amplifier without coupling among the channels.

How the amplifier is used in this connection is shown in figure 20, where the numbers refer to the crosspoint connections.

Writing a nodal equation for the grid of one of these amplifiers gives

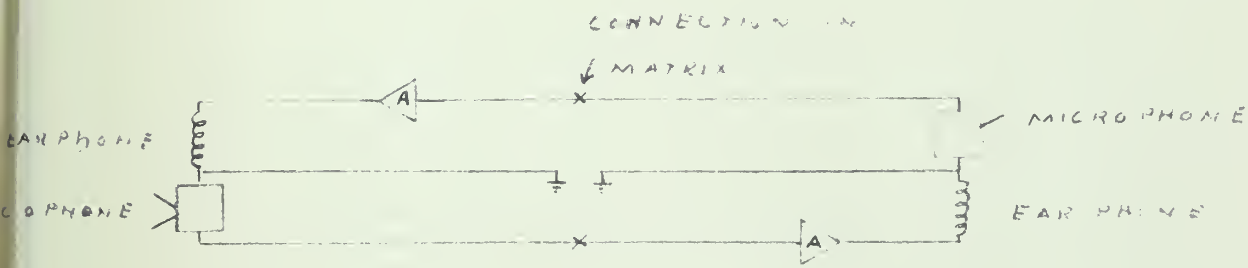
$$\left(\frac{X_o}{A} - X_1\right)\frac{1}{R_1} + \left(\frac{X_o}{A} - X_2\right)\frac{1}{R_2} + \cdots + \left(\frac{X_o}{A} - X_N\right)\frac{1}{R_N} + \left(\frac{X_o}{A} - X_o\right)\frac{1}{R_o} = 0$$

where $x_1, x_2, \cdots x_n$ Input voltages x_o - Output voltage

$R_{1_1}, R_{2_1}, \cdots R_n$ Input resistances

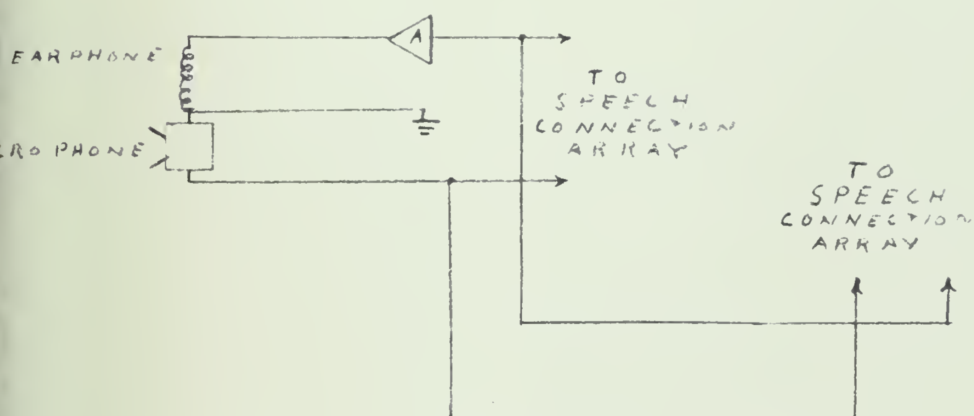
FIG 19

FIG 19a



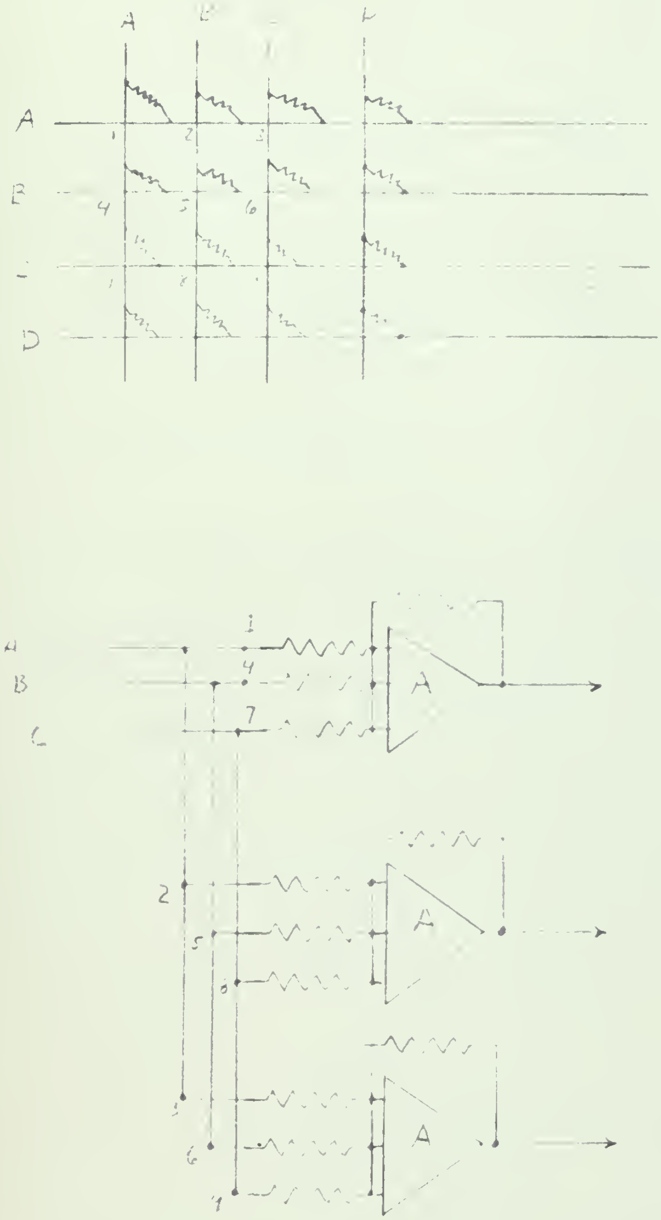
SPEECH CHANNEL CONNECTION

FIG 19b



SPEECH CHANNEL INPUTS TO MATRIX

FIG 20



SUMMATION AUDIO AMPLIFIER CONNECTION

rearranging

$$X_o = -\left(\frac{1 - 1}{1 - AB}\right) R_o \left(\frac{X_1}{R_1} \quad \frac{X_2}{R_2}\right) \qquad \frac{X_n}{12_n}$$

where $$\beta = \left[1 + \frac{R_o}{R_1} \quad \frac{R_0}{R_2} \qquad \frac{R_o}{R_n}\right]^{-1}$$

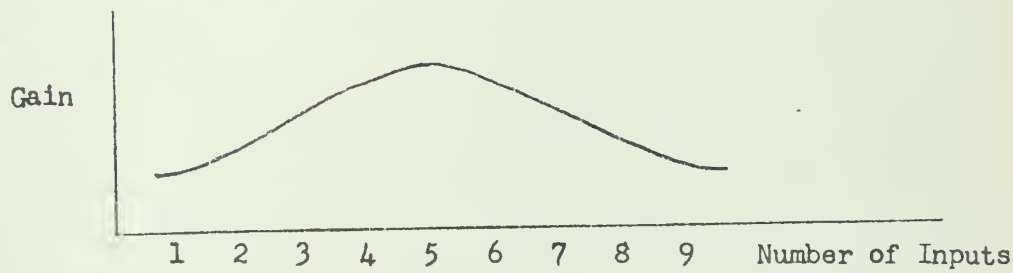
this may be written

$$X_o = -R_o \left[\frac{X_1}{R_1} + \frac{X_1}{R_2} + \cdots + \frac{X_n}{12n}\right] \qquad R_1 = R_2 = \cdots = R_n$$

Assuming $|A| \gg 1$

$$X_o = -R_o \sum_{i=1}^{n} \frac{1}{R_i}$$

Thus the gain is dependent upon the number of resistors involved in the input, but it was stated earlier that the inputs to the amplifier are variable from one to N, where conceivably N could be as large as 60, but is highly unlikely. Therefore, to properly choose the resistors for the circuit it must be designed for the average number of inputs, and sufficient gain built in to compensate for the decrease in gain on each side of this curve is



To determine what gain should be built into the amplifier the entire audio channel must be considered.

Telephone practice[18] states that the sum of the losses in two loops along or two loops plus the maximum trunk losses should not exceed about 30 db. Approximately 6 db loss should be allowed for each loop; thus on an internal connection there should be allowed 12 db of loss in the wiring

and connections. On external lines there is this 12 db of loss plus 3-6 db of loss if the signal must pass through a hybrid transformer somewhere in its path. The drop-off in gain, if the number of inputs to the amplifier should be considered, plus the insertion loss of the pad at the station panel to enable the operator to control the volume level, can be estimated at 3 db. This is a total of 15 db +(3 to 6) db on external connections. Therefore, a gain of 20 db should be designed into each amplifier.

The same scheme of amplification is used for the monitor amplifier, but here the average number of inputs may be considered as two, as it is not considered feasible for an operator to monitor more than two channels at any one time.

7. UHF Radio and Channel Selection.

Normally the UHF radios used in this system will be preset on channels. A station desiring to transmit to aircraft on a preset channel will switch this headset to that channel in the switching-central, and commence transmission. The UHF panel to accomplish this is shown in figure 8. There are only 7 buttons being used but 8 code slots are reserved for UHF, thus one more unit may be added if desired.

Operational studies[2] indicate that more than 6 UHF receiver transmitters will seldom be required. Therefore, capability is provided for 6 preset channels plus Guard.

Although the receiver-transmitters are time-shared by operators in the system, no special provision is provided to indicate what receiver transmitters are busy. Here again advantage is taken of the operator's intelligence and he is required to listen before transmitting to insure that no one else is using the channel.

Although it is stated in the requirements that capability should be provided to change channels on a selected receiver-transmitter, this capability would be recuired by few operators and its addition as an integral part of the system would be a needless complication to the system. It would be desirable to provide this function as an optional part of the system which can be added whenever it is required. This is the approach taken in the design of this phase of the system. It is designed as a completely separate function so that the addition or deletion of it will not affect the rest of the system. Therefore, the design of the channel selection will be discussed in this section.

The problem in channel selection is to cause any one of 21 events to occur, that is, the selection of one of 20 preset channels in addition to

a guard channel. The conventional solution is to use a 20-contact rotary switch to step the channels at the receiver-transmitter. However, this technique would require a minimum of 21 wires, which is considered to be excessive wiring. Consequently another approach is desired. The following approach is proposed as the desired solution.

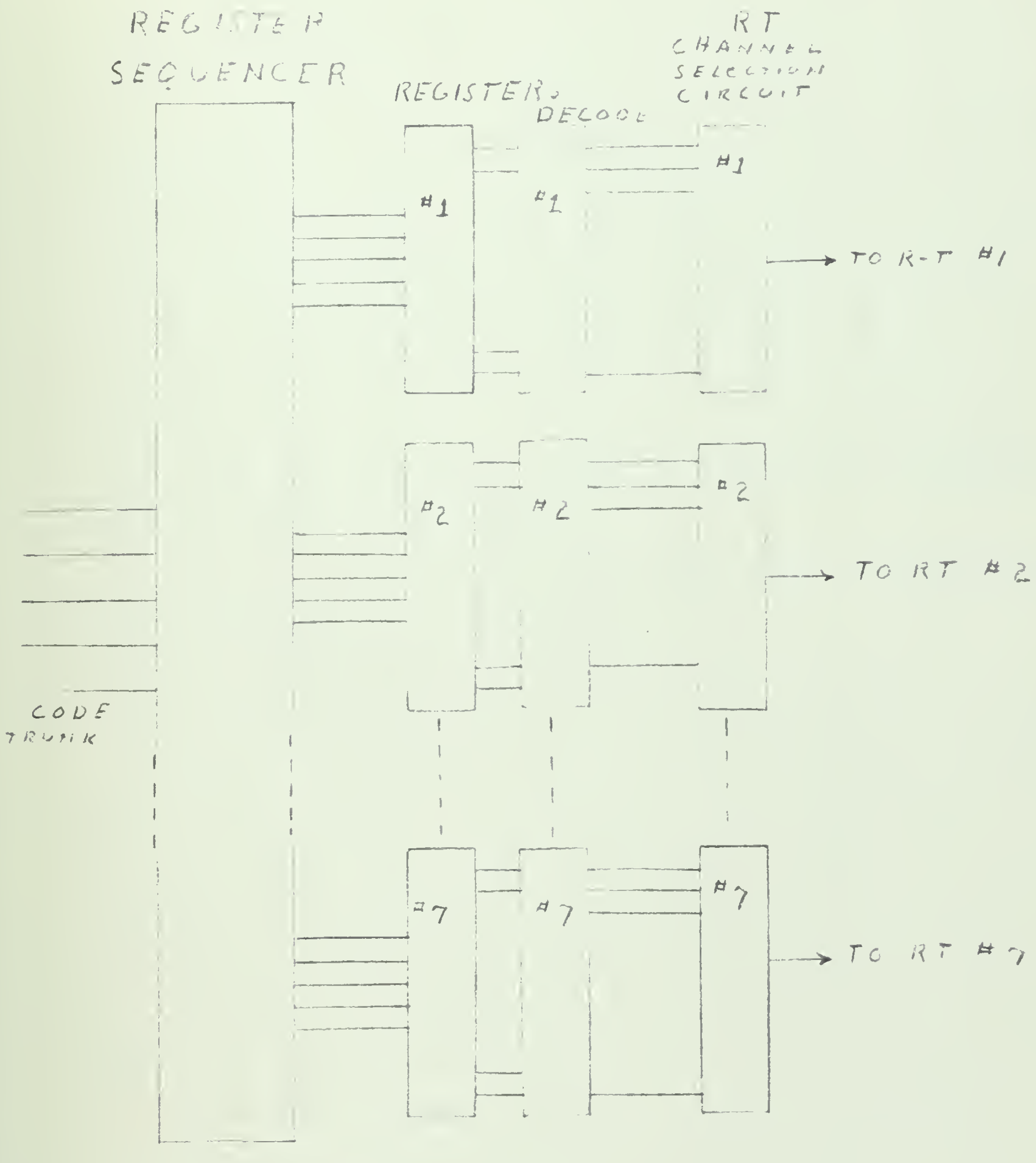
1. To describe a channel uniquely, a binary code can be used. Using a serial coding technique the absolute minimum of two wires would be required. However, this technique is objectionable from the standpoint of simplicity; consequently, a parallel code will be utilized. Since there are 21 events to be described, a five digit code is required. This allows 32 possible actions, of which 21 are required and 00000 is not used. Thus 10 spare codes are available as a desirable by-product, for more channels may be made selectable if desired. The five digit code is obtained by using a five wafer 20 position rotary switch and letting missing contacts indicate binary "0", since that will put no D.C. on code wires.

2. To decode the channel selection code at the receiver-transmitter and to meet the requirement that no two receiver-transmitters be tuned up on the same channel at the same time, the following solution was devised. The solution is shown in figure 21 in block diagram form. The channel selection code trunk is run separately from the switching central code trunk. The RT channel selection circuit is a servo operated device which steps the receiver-transmitter to the proper channel depending upon which wire has voltage placed upon it by the register decode circuit.

a. A register and associated decoding circuit are provided for each transmitter-receiver.

b. When the channel selection switch is moved to a particular channel all five registers are sampled sequentially by a sequencer

FIG 21



UHF CHANNEL SELECTION

circuit one through five. If the code present on the code wires is present in a register the sampling stops and the light under the receiver-transmitter button in which the code is present is illuminated. If no correspondence is obtained, the sequence is repeated until the first empty register is found, the sampling then stops and the code is fed into the empty register. The light under that receiver-transmitter button is illuminated, indicating that the channel is set up and which button or receiver transmitter is to be pressed.

The solution proposed in (b) imposes two additional requirements on the channel selection as follows:

a. The channel selectors puts no code on the code wires until it has been rotated to the desired channel.

b. No two or more operators try to change channels at the same time.

These requirements may be met by using a spring loaded normally up channel selection switch. It is rotated to the desired channel and pushed down. This places the D-C code on the wires and lights a light integral to the switch. This light informs all other operators that channel selection is in progress so that no other operator will attempt to change channels. The light is turned off when the channel selection sequencer stops on a register. When the operator releases the channel selection switch it returns to its up position removing the code from the wires.

The code is retained in the register as long as the operator is using UHF unit. When he is finished he pushes the off button associated with that unit which clears the register and turns off the light on the panel thus enabling the unit to be switched to a new channel.

For guard channel selection the above procedure must be modified. In the previous procedure if a channel was not already set up and no

empty register existed, the channel desired could not be set up. This situation cannot be allowed to exist for the guard channel as it is an emergency channel and must be supplied on demand. To meet this situation the guard channel code is selected as 11111. Now when this code is present and the situation of neither a channel already set up nor no empty register exists, the sequencer stops on the last register and overrides the code present and switches the receiver-transmitter to guard.

Although this is probably not the best solution to this problem, it is considered a feasible solution since normal UHF operation will be via preset channels. The capability to change channels would be provided to a limited number of operators. Although figure 21 shows registers for all seven receiver-transmitters, only two or three units would actually be given channel changing circuitry. The remainder would be left on preset channels. When these factors are considered the solution is a feasible one.

8. Selection of Administrative Communications Switching.

In this portion of the system requirements are much less demanding. The time lag of conventional communication techniques can easily be tolerated since this portion is destined to serve only administrative and maintenance personnel. It might be argued that when equipment fails quick action is needed to put it back into operation. This is true, but the demand for quick operation is placed upon the technician, not the communication system serving the technician. In addition it is wasteful to build complexity into a system where complexity is not really required.

In view of all these factors the solution appears to be the use of a manual switchboard to provide the communication of administrative and maintenance stations. The use of a manual switchboard in this system adds several desirable features. The most important is that it adds a highly desirable back-up means of communication to the automatic switching central. No system of communication should be complete in itself in the military environment, for there should always be an alternate means of communication available in case the primary system fails, or is rendered inoperative by some means such as enemy action. The use of a switchboard further extends the flexibility of the system. Channels of communication not available through the switching central may be obtained through the manual board. This alleviates the necessity for providing for every possible communication demand in the automatic system. Those channels for which a low probability of demand exists and those which are "nice to have but not really necessary" may be obtained through normal means of communication.

Special means will have to be provided at the switchboard to enable the stations on the automatic switching system to signal the operator that

a demand for service exists. This may be done by detecting the 1000 cycle ringing used by the automatic system and using this for signaling. The 1000 cycle could be converted to the normal 20 cycle AC ringing employed by the usual common battery telephone system and used to signal the operator. Either method may be used, but the second method has the advantage that such means already exist, so is easily supplied.

Considering all these factors the solution adopted for the administrative communication is the use of a manually operated switchboard.

9. System Evaluation.

With the completion of the last phase of the system design the entire system operation should be reviewed to determine how well it meets the requirements laid down in part 2. A schematic of the overall system is shown in Figure 22. This schematic shows primarily the control and signalling functioning of the system. Several circuit details are omitted in the interest of clarity and the audio channels are shown only in outline. Using this diagram and the illustrations of circuit functioning in parts 4, 5, and 6, the system functioning for several types of calls such as internal, external and monitoring may be traced.

The first operation to be traced is that of two internal or local stations A and B where station A desires to talk with station B. The sequence of operations by the operators and the system is as follows:

1. The operator at station A selects the station and "talk" button associated with station B on his panel and presses it. This puts a binary code representing station B in the panel register. On station A's panel the light for station B is turned on.

2. At station A's designated time period a service pulse supplied by the local counter allows the binary code to be read out to the code trunk and transferred to the switching central.

3. The switching central decodes the binary code, notes that a talk connection is desired, and places a voltage on station B's wire in the talk matrix.

4. The central counter places a voltage on station A's wire since the time period belongs to station A. The combination of these two events places a double voltage at the A-B crosspoint firing the associated gas tube.

5. Since this is the first selection of two crosspoints (AB and BA) no speech connection is made (see figure 15a).

6. Meanwhile at station B's panel the fact that its binary code is on the code wires is noted by an output from the station "and" gate.

7. This output, with the output from the counter indicating which station's time slot is on the wire, is combined to illuminate in a blinking mode station A's light on station B's panel.

8. Station B, noticing the blinking light denoting the fact that Station A desires to talk with him, pushes the two buttons associated with station A which enters stationA's code in the panel register and switches station A's light to steady illumination.

9. At station B's designated time slot the local counter supplies a service pulse sending out the binary code stored in the register.

10. This code is decoded at the switching central and the B-A crosspoint is selected firing the associated gas tube.

11. Since this is the second selection the relay is activated and the speech connection closed. Conversation may now begin.

12. At the end of the conversation, since station A was the originator of the call, it presses the two buttons associated with station B, setting station B's code in the panel register.

13. This binary code is sent out during station A's designated time slot.

14. The code is decoded at the switching central and activates station B's wire which, in conjunction with the station A wire made "hot" by the counter, selects the A-B crosspoint.

15. This selection turns off both gas tubes, opening the speech connection.

The procedure for calling station B when station B is an external station is as follows:

1. Steps one through four are the same as above.

5. Since this is an external station, the speech connection is made at the switching central.

6. The D-C code is converted to tones and sent via the speech channel to the remote station's system.

7. The tones are detected and the binary code stored in a register.

8. The register is serviced precisely as if it were a station on the system and the code placed on the code wires. The switching central will interpret this code and activate the A-B crosspoint but nothing is connected to this crosspoint since the speech connection logic is only two-state for external stations.

9. Steps 9 through 13 are the same as steps 5 through 9 above.

14. This selection fires the associated gas tube closing the speech connection in the remote system.

15. Now the speech connection is complete at both ends of the channel so conversation may begin.

16. At the end of conversation both stations must repeat initial calling action to break the speech connection.

To establish a monitor connection, the procedure is as follows:

1. The station operator selects the station and monitor buttons associated with the station or channel that it desires to monitor. The buttons are pressed putting the code in the panel register.

2. At the designated time slot for the station the code is sent out on the code trunk.

3. At the switching central the fact that a monitor connection is

noted by the state of the seventh code wire. The selection circuitry is switched to the monitor matrix where the desired crosspoint is selected as indicated previously.

4. The associated gas tube is fired closing the monitor connection since this is a two-state connection.

5. To disestablish the monitor connection, steps one through four are repeated.

6. The associated gas tube is deionized opening the speech connection.

From the above it is seen that placement of a call is a simple rapid procedure.

The control circuitry is designed for the maximum number of stations but it can handle any number of stations from two up to the maximum number of 62 stations. This flexibility is the strongest attribute of the switching network. As noted earlier, stations may be added as desired by supplying a station panel, running the necessary number of wires and plugging in the required speech connection relay circuits in the switching central. By use of this "plug-in" and common code trunk principle, capabilities of speech and monitor connection may be added at will to meet any desired operation.

The logic circuitry employed throughout the design is simple and straightforward. The number of levels of logic is restricted to two and the speed of operation is on the order of milliseconds, both of which contribute to ease of circuit design. All logic is semiconductor circuitry so may be built on plug-in cards which provide ease of maintenance.

Perhaps the greatest disadvantage of the system is the number of wires required for each station panel. From figure 22 it is seen that each panel requires the following:

A 1. Four wires for the voice channel;

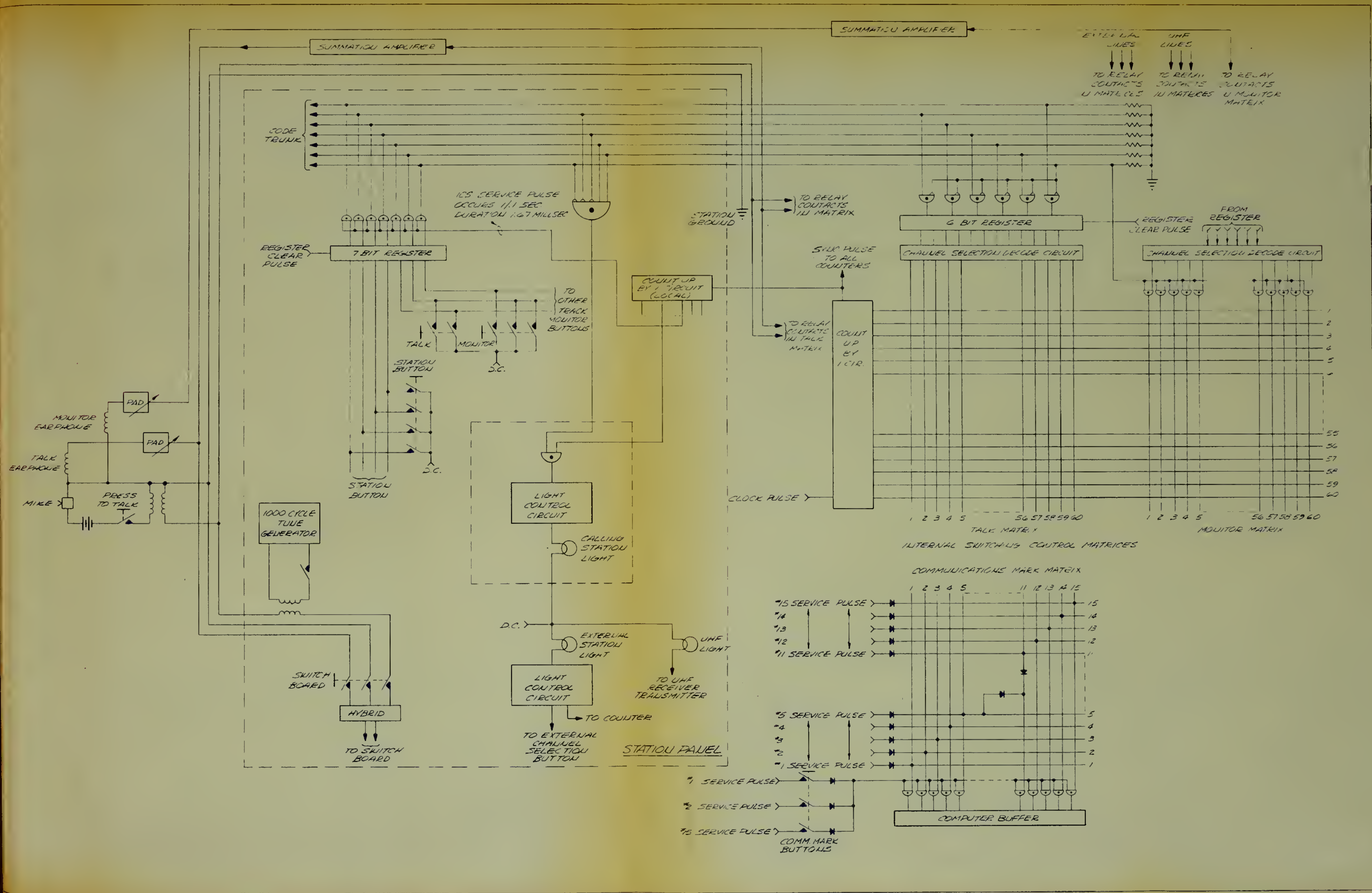
2. Seven wires for the code trunk;

3. One wire for counter synchronization.

A total of 12 wires is required to provide an output from each panel. This is no great disadvantage when used in a situation such as M.T.D.S. where cables are available to carry the wiring or in a situation where wiring runs would be short on the order of 100 yards or less. However, it would be very real disadvantage in a dispersed situation where station locations are rather far apart. To cope with this situation an alternate solution is proposed.

The same basic scheme would be employed of a common code trunk. The trunk would be created by a phantom circuit[18] utilizing the monitor-talk channels of the station's headset. The D-C binary code would be converted to tones at the station panel and sent to the switching central and all other stations via this common, phantom trunk. In addition, the synchronizing pulses for the local counter could also be sent via this trunk since frequency division as well as time division would be employed. Employment of this scheme would involve considerable use of ringing circuits and filters but it is entirely feasible and would reduce wiring requirements to a minimum of four wires. These are the wires necessary to carry the monitor and speech channels.

Its use was not considered feasible in this design as in general it was considered more economical and simpler to use wiring than additional circuitry, since cable runs were short in the operational environment for which this solution is proposed.



SUMMATION AMPLIFIER
SUMMATION AMPLIFIER
EXTERNAL LINES
UHF LINES
TO RELAY CONTACTS IN MATRICES
TO RELAY CONTACTS IN MATRICES
TO RELAY CONTACTS IN MONITOR MATRIX
CODE TRUNK
ICS SERVICE PULSE OCCURS 1/1 SEC DURATION 1.67 MILLISEC
STATION GROUND
TO RELAY CONTACTS IN MATRIX
6 BIT REGISTER
REGISTER CLEAR PULSE
FROM REGISTER
REGISTER CLEAR PULSE
7 BIT REGISTER
CHANNEL SELECTION DECODE CIRCUIT
CHANNEL SELECTION DECODE CIRCUIT
SYNC PULSE TO ALL COUNTERS
COUNT UP BY 1 CIRCUIT (LOCAL)
TO OTHER TRACK MONITOR BUTTONS
TALK
MONITOR
D.C.
TO RELAY CONTACTS IN TALK MATRIX
COUNT UP BY 1 CIR.
STATION BUTTON
PAD
PAD
MONITOR EARPHONE
TALK EARPHONE
MIKE
PRESS TO TALK
D.C.
STATION BUTTON
1000 CYCLE TONE GENERATOR
LIGHT CONTROL CIRCUIT
CALLING STATION LIGHT
CLOCK PULSE
1 2 3 4 5 56 57 58 59 60
TALK MATRIX
1 2 3 4 5 56 57 58 59 60
MONITOR MATRIX
INTERNAL SWITCHING CONTROL MATRICES
COMMUNICATIONS MARK MATRIX
#15 SERVICE PULSE
#14
#13
#12
#11 SERVICE PULSE
D.C.
EXTERNAL STATION LIGHT
UHF LIGHT
TO UHF RECEIVER TRANSMITTER
SWITCH BOARD
HYBRID
TO SWITCH BOARD
LIGHT CONTROL CIRCUIT
TO COUNTER
TO EXTERNAL CHANNEL SELECTION BUTTON
STATION PANEL
#5 SERVICE PULSE
#4
#3
#2
#1 SERVICE PULSE
#1 SERVICE PULSE
#2 SERVICE PULSE
#15 SERVICE PULSE
COMM MARK BUTTONS
COMPUTER BUFFER

10. Conclusions.

In this paper a proposed design of an automatic communications switching network has been presented. The design philosophy of such a switching network was discussed. The primary emphasis in the system was reduction of the time required to obtain a communication channel to another station and the meeting of all system requirements. An equally important emphasis was placed upon reliability, and simplicity of design. The reduction of the time required to acquire a channel was achieved by use of semiconductor information processing circuitry, which affords an increase in operating speed without a great increase in weight and volume.

The logic circuitry used for information processing is of a general purpose design and arranged in building blocks. This affords great flexibility in implementing required system functions. New stations may be added as desired up to certain limitations and the station capabilities of monitoring and obtaining speech channels may be established as desired. In view of all these factors and considering that the system meets in full or exceeds all the requirements established in part 2, the system is considered to be a workable, practical communications switching system.

Since this system, as presented, exists only on paper undoubtedly there are many improvements that suggest themselves, particularly during the development of such a switching system. Particularly there are areas which would bear further investigation to enable the system to handle more communications channels and stations, and to extend its flexibility.

Recommendations for such improvements in an automatic switching system to increase its capabilities and functions are:

1. Use of voice frequency tones using the speech channel to replace the present code trunk.

2. The separation of the built-in memory functions into a separate bulk memory for use in controlling the switching in the central.

3. The use of voice frequency gas tubes to replace the relays presently used in the central.

4. An optimum method for keying a transmitter remotely via the speech channel to the transmitter.

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APPENDIX I

INDUCTIVE INTERFERENCE OR CROSSTALK

Interference from paralleling circuitry into a communication line is usually regarded as noise whether from a power line, or circuitry carrying frequencies in the voice range. If the interference is from another communication channel, it is called crosstalk. Experiments have shown[18] that if the speech on the line is at least 13db higher than the interfering source or crosstalk, there will be no difficulty in distinguishing the conversation from the noise. Thus it is important that any communications employing voice frequency over wires be as free from crosstalk as possible.

To produce this interfering noise, energy must be coupled from one circuit to another. This transfer of energy may be accomplished in three ways:

1) Through leakage or conducting paths between the two circuits.
2) Coupling via the magnetic fields established by the currents flowing in the two circuits.
3) Coupling via the electric field established by the voltages or electric charge present on the wires.

The influence these factors have upon the choice between a single wire with ground return communications channel and a channel employing a metallic return, or two-wire system, will be shown by the following analysis.

The mutual impedance between two circuits will determine how much energy from one circuit is coupled into the other. Consequently, examining the mutual impedance between an interfering source and

a) a single wire with ground return channel
b) a two-wire system

will provide a valuable piece of information for evaluating the choice between these two channels. Therefore, we proceed to derive the mutual impedance for these two cases.

Assume:

1) The interfering source to be a single wire with ground return carrying a frequency of 1000 cycles.

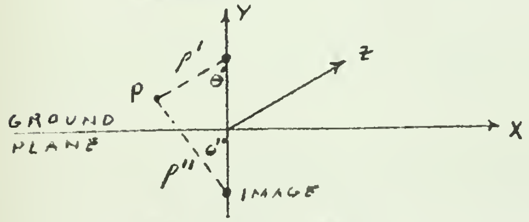
2) The communication channel wire to be at the same height with:

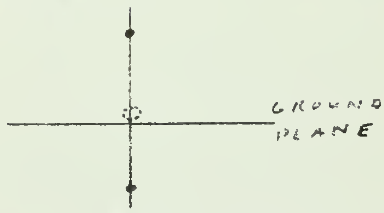
   a) a ground return

   b) a metallic return path carrying an equal and opposite current.

3) Perfectly conducting ground.

The geometry of the two cases is as follows:





A formula for the mutual impedance (Z_{12}) between two parallel ground return circuits with wires at a height (h) above ground and a separation (X) between their vertical planes has been derived by J. R. Carson.[19] Since the derivation is complete and straightforward, it is not given here, and only the results of the derivation are utilized.

The mutual impedance (Z_{12}) is:

$$Z_{12} = 1Z\ W\ \mathrm{LoG}\ (P^{II}/_{P}I) + 4W \int_{0}^{\infty} \left[\left[(u^2 + i\right]^{\frac{1}{2}} - u \right] e - h^1{}_1 + h^1{}_2)u \cos X'u\ du$$

where

$$P^{II} = \left[(h_1 + h_2)^2 + X^2\right]^{\frac{1}{2}} = \left[4h^2 + X^2\right]^{\frac{1}{2}} \qquad \text{For } h_1 = h_2$$

$$P^{I} = \left[(h_1 - h_2)^2 + X^2\right]^{\frac{1}{2}} = X \qquad \text{For } h_1 = h_2$$

$$h^I{}_I = h_1 \left[\alpha\right]^{\frac{1}{2}} \qquad h^1{}_2 = h_2 \left[\alpha\right]^{\frac{1}{2}} \qquad X = X\left[\alpha\right]^{\frac{1}{2}}$$

$\left[\alpha\right]^{\frac{1}{2}} = 4\pi\lambda\omega$ $\quad\lambda$ = conductivity of ground in elm. cgs units

$$j = \left[-1\right]^{\frac{1}{2}}$$

$$Z_{12} = Z^{o}{}_{12} + Z^{1}{}_{12}$$

The calculation of the mutual impedance depends on the evaluation of an infinite integral of the form.

$$J\ (P\ Q) = \int J = \int_0^\infty \left[(u^2 + i \right]^{\frac{1}{2}} - u)\ e^{-Pu} \cos q\ u du$$

Fortunately this integral has been solved and the solution is in the form of two series.

$J = P + jQ$ For the case $r \leq .25$ $\gamma = \left[P^2 + Q^2\right]^{\frac{1}{2}}$ $\theta = \mathrm{TAN}^{-1} \frac{P}{Q}$

$$P = \frac{\pi}{8} - \frac{1}{3\sqrt{2}} r \cos \theta + \frac{r^2}{16} \cos 2\theta \quad (.6728 + \mathrm{LoG}\ \frac{2}{r}) + \frac{r^2}{16}\ \theta \sin z\ \theta$$

$$Q = -0.0386 + \mathrm{LoG}\ (\frac{2}{r}) + \frac{1}{3\sqrt{2}} r \mathrm{CoS}\ \theta$$

The series takes other forms for $r > .25$, but these are of no importance here, so are not included.

$Z_{12} = Z^o_{12} + {}^4wJ$ where J is given above, and

$$r = \left|\alpha\right|^{\frac{1}{2}} \left[(h_1 + h_2)^2 + x^2\right]^{\frac{1}{2}} = P^{11} \left[\alpha\right]^{\frac{1}{2}}$$

$$\theta = \sin^{-1} (\frac{x}{p^{11}})$$

Let $f = 1000$ cycles $h_1 = h_2 = 3$ cm

$\alpha = 10^{-12}$ $x = 3$ cm

$$\sqrt{\alpha} = \left[4\pi 2w\right]^{\frac{1}{2}} = \left[8\pi^2\ 10^{-12}\ 10^3\right] = \left[79.0\ 10^{-9}\right]^{\frac{1}{2}}$$

$$\sqrt{\alpha} = 2.81.10^{-4}$$

$$r = \sqrt{\alpha} \left|(2h)^2 - x^2\right|^{\frac{1}{2}} = 2.81.10^4 \left[36 - 9\right]^{\frac{1}{2}} = 2.81.10^{-4} \times 5.2$$

$$r = 14.6.10^{-4}$$

$$\theta = \sin^{-1} \frac{3}{5.2} = .576 = 35.3^{o}$$

$$P = \frac{\pi}{8} - \frac{14.6.10^{-4}}{3\ 2} \times .816$$

$$= .392$$

$$Q = -0.0386 + \tfrac{1}{2}\ \mathrm{LoG}\ \left(\frac{2}{14.6.10^{-4}}\right)$$

$$= -0.0383 + \mathrm{LoG}\ 1370 = -0.0383 + 1.568$$

$$= 1.5297$$

Therefore the mutual impedance for the circuit employing ground return is

$$Z_{12} = P + \dot{j}q = 4w\ (.392 + j1.5297)$$

Now consider the case of a metallic return path, lying at the surface of the ground for simplicity. This introduces a new mutual impedance between the interfering source and the return path which must be subtracted from the mutual impedance derived above.

For this case:

$$P^{11} = \left[h^2 + x^2\right]^{\frac{1}{2}} = \left[9 + 9\right]^{\frac{1}{2}}$$

$$= 4.24$$

$$\theta = \frac{3}{4.24} = .707 = 45^{o}$$

$$r = 4.24\sqrt{2} = 4.24 \times 2.81.10^{-4}$$

$$= 11.9.10^{-4}$$

$$P = \frac{\pi}{8} - \frac{11.9.10^{-4}}{3\ 2}\ .707 = .393 - 1.98.10^{-4}$$

$$= .391$$

$$Q = -0.0386 + \tfrac{1}{2}\ \mathrm{LoG} \frac{2}{11.9.10^{-4}} + 1.98.10^{-4}$$

$$= -0.0382 + 1.613$$

$$Q = 1.5748$$

Therefore the resultant mutual impedance between the interfering source and the parallel two-wire communication channel is

$$Z^1_{12} = 4W\left[.392 - .382 + j(1.5297 - 1.5748)\right.$$
$$= 4W \quad (01 - j\ .0451)$$

Thus it is apparent that the presence of a metallic return gives a striking reduction in mutual impedance as may be expected. Although the result is based upon rather ideal conditions of perfectly conducting grounds, equal currents, etc., it is clear that if the mutual impedance in the second case was ten times as bad, the resultant impedance would still be considerably less than the case for a single wire with ground return.

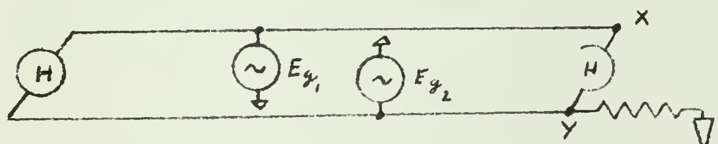
Before a choice between the two circuits or channels is made, let us further examine the two-wire case for the three conditions of

a) leakage

b) coupling via magnetic field

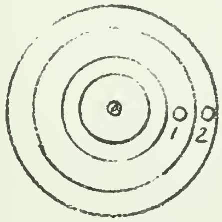
c) coupling via electric field

1. A leakage path causes unbalance to ground as shown

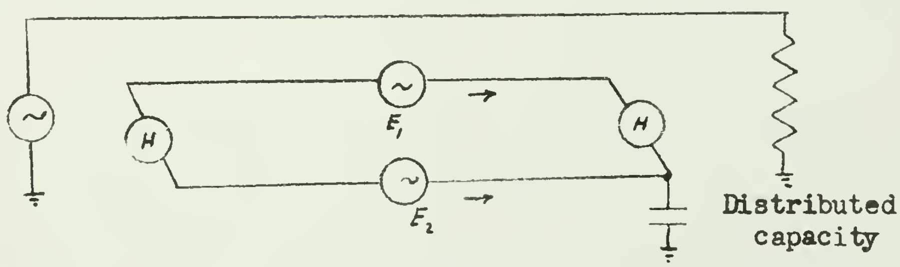


This unbalance causes noise currents to flow, since now the point Y is at a lower potential than point X, thus causing noise currents to flow even if $E_{g_1} = E_{g_2}$.

2. Current flowing in the wire of the interfering source will set up a magnetic field which will link the two-wire communication channel as shown:



Since line 1 is closer to the interfering source than line 2, a higher voltage will be induced in line 1, thus causing a difference in potential between line 1 and line 2. This induced voltage is in series with the line as shown where the induced voltages are represented by generators E_1 and E_2.



The difference in potential caused by $E_1 > E_2$ will cause a current to flow in the circuit which will vary at the same rate as the interfering source, thus causing crosstalk and noise in the headset (H).

The coupling via the electric field can be examined by considering the voltages and distributed capacitances between the two circuits.

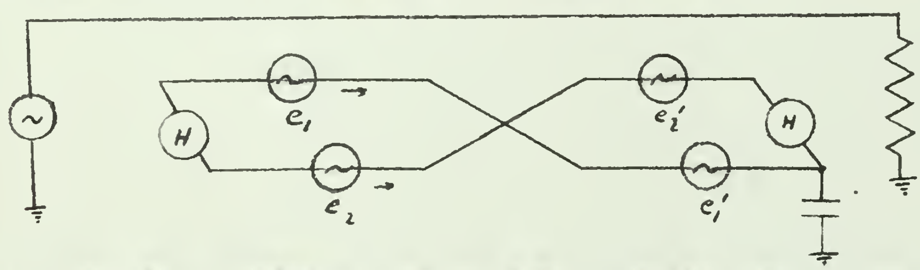
The effective voltage E between the interfering source and ground will by voltage divider action divide inversely proportional to the capacitances as follows:

$$E_{g_1} = \frac{EC_1}{C_1 + C_{g_1}} \qquad E_{g_2} = \frac{EC_2}{C_2 + C_{g_2}}$$

and $E_{g_1} > E_{g_2}$ due to the distances between wires.

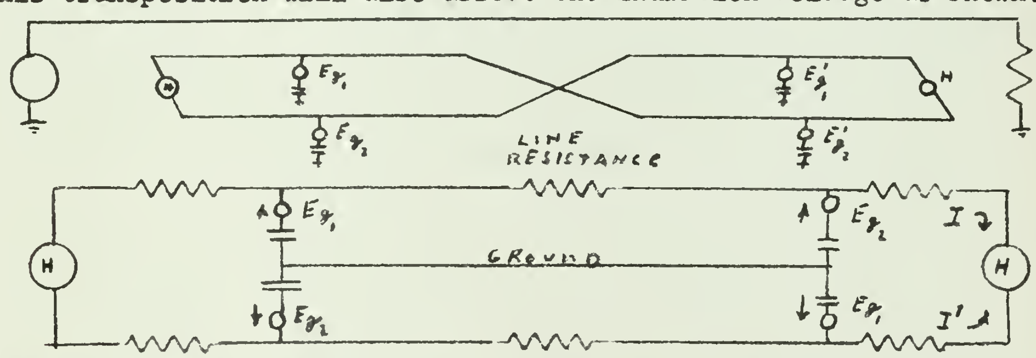
Since $E_{g_1} > E_{g_2}$ there is a potential difference and noise current will flow.

Thus it is apparent that the two-wire system is still highly susceptible to interference and crosstalk. However, with a two-wire system there is an effective method of correcting the induced voltages in the system by a process known as transposition. The transposition of a wire line will tend to balance it with respect to the interfering course as shown:



Now, since the lines occupy the same relative position with respect to the interfering source $e_1 + e^1_1 = e_2 + e^1_2$ and the voltages induced in series by the magnetic field coupling are nearly equalized and tend to cancel each other.

This transposition will also affect the induction voltage as shown:



Analyzing the equivalent circuit shows that voltage $E^1_g - E^1_{g2}$ will tend to send a noise current I^1 up through the headset on the right and voltage $E_{g_1} - E_{g_2}$ will send a current I down through the same headset. Thus I and I^1 will tend to cancel each other.

A single transposition is not effective in reducing crosstalk due to the phase changes of the signals in the two channels. Since crosstalk is an induced effect, its instantaneous value in a small section of line depends upon the position of the section with respect to the cycle of current in the interfering circuit. Therefore, it is necessary that the transposed section be short compared with the wavelength in the interfering circuit.

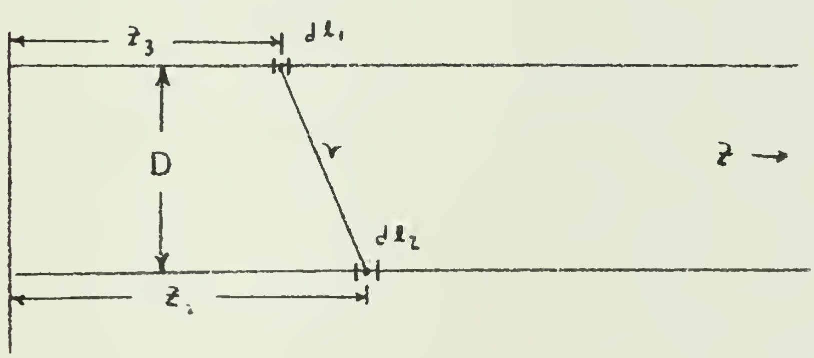
Thus it can be seen that transposition is an effective method of reducing interference or crosstalk due to coupling by the magnetic and electrostatic fields. This is the second great advantage of a two-wire over a single wire system--that it can be transposed while a single wire cannot.

As stated earlier the threshold of intelligibility for crosstalk is 13 db. Therefore it is of interest to see if the single wire circuit crosstalk exceeds this value.

I. Mutual Impedance

A formula for the mutual impedance of two parallel wires may be derived which will be more useful for this purpose than the previous formula.

Consider two parallel wires as follows:



$$d\ell_1 = dz_1$$

$$r = \left[D^2 + (z_2 - z_1)^2 \right]^{1/2} .$$

VECTOR POTENTIAL A POINT DUE TO CURRENT

ELEMENT $I_1\, dl_1$

$$A_z = \frac{\mu_0 I_1}{4\pi} \int_0^l \frac{dl_1}{r}$$ SUBSTITUTE FOR dl_1 AND r

$$A_z = \frac{I_1 \mu_0}{4\pi} \left\{ \int_0^{z_2} \frac{dz_1}{\left[D^2 + (z_2 - z_1)^2\right]^{1/2}} + \int_{z_2}^{l} \frac{dz_1}{\left[D^2 + (z_1 - z_2)^2\right]^{1/2}} \right\}$$

$$A_z = \frac{\mu_0 I_1}{4\pi}\, 2\, Ln\left(z_2 + \sqrt{D^2 + z_2^2}\right) + 2\, LN\left[(L - z_2) + \sqrt{D^2 + L - z_2^2}\right] - 2\, LN\, D$$

ELECTRIC FIELD STRENGTH AT POINT P IS

$$\bar{E} = -\frac{\partial A}{\partial t}$$

$$E_z = -\frac{\mu_0\, dI_1}{2\pi\, dt} \left\{ LN\left(z_2 + D^2 + \sqrt{D^2 + z_1^2}\right.\right. + 2\, LN\left[L - z_2 + \sqrt{D^2 + L - z_2}\right] - 2\, LN\, D$$

EMF INDUCED IN WIRE 2

$$V_2 = \int_0^l E_z\, dz = -$$

$$= \frac{\mu_0 L\, dI_1}{2\pi\, dt} \left\{ LN\left[\frac{L}{D} + \sqrt{1 + \left(\frac{L}{D}\right)^2} - \sqrt{1 + \left(\frac{D}{L}\right)^2} + \frac{L}{D} \right\}\right.$$

If $L \gg D \quad \therefore \quad \frac{L}{D} \gg 1$

$$V_2 \approx \frac{\mu_0\, L\, dI_1}{2\pi\, dt}\, LN\left(\frac{2L}{D} - 1\right)$$

$\mu_0 = \frac{1}{4\pi} \qquad LN = LOG_2$

Now $M = \frac{E_2}{I}$ so M may be written as:

$M = .00508 \quad 2.303 \, LOG_{10} \frac{2}{D} - 1$ microhenrys

The situation assumed is that there are 10 wires in a cable with a common ground in same cable

ℓ --length of wire

D--spacing between wires

Let = 100 ft.

D	$3 + LOG_{10} \frac{2.4}{D}$	$2.303 \left[3 + LOG_{10} \frac{2.4}{D}\right] - 1$	M u h	$E_{Induced}$	$E_{Induced}$
.1	4.38	9.1	55.5	1.56	2.44
.2	4.08	8.4	51.2	1.44	2.07
.3	3.91	8.0	48.8	1.37	1.88
.4	3.78	7.71	47.2	1.32	1.74
.5	3.68	7.48	45.6	1.28	1.64
.6	3.6	7.3	44.5	1.25	1.56
.7	3.54	7.15	43.6	1.23	1.51
.8	3.48	7.03	42.9	1.21	1.46
.9	3.43	6.9	42.1	1.18	1.39
1.0	3.38	6.8	41.5	1.17	1.37
					17.06

$$E_{induced\ total} = E_1^2 \times E_2^2 - - E_{10}^2$$

$$= 17.06$$

$$= 4.13 \text{ millivolts}$$

$$db = 20 \text{ Log } \frac{4.47}{4.1316^{-3}} = 20 \, (3 + .036)$$

$$= -60.6 \text{ db}$$

So this effect may be neglected.

II Capacity between two parallel wires in the presence of ground

$$C = \frac{3.677\,\ell}{\log_{10} \frac{2D}{d}} \quad \text{micro micro farad per foot}$$

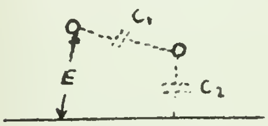
ℓ = length of wire

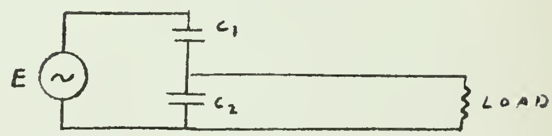
D = spacing between wires

d = diameter of wire

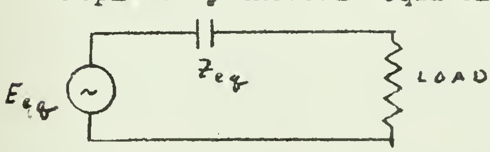
Assume: 1. #26 wire Dia. = .016 inch
2. wires are average .5 inches above ground: C_2 = 240 u u per 100 feet

the situation is:





Replace by thevenin equivalent circuit to get true induced VOLTS



D inches	125 D	$\log_{10} \frac{2D}{d}$	C_1	$C_1 + C_2$	$E = \frac{E\,C_1}{C_1 + C_2}$	Zeg Kilo ohms	$E_{induced}$ VOLTS
.1	12.5	1.097	335	575	2.6	9.25	.40
.2	25.0	1.38	266	506	2.34	10.5	.39
.3	37.5	1.57	234	474	2.21	11.2	.38
.4	50.0	1.699	216	456	2.11	11.6	.36
.5	62.5	1.79	205	445	2.06	11.93	.36
.6	75.0	1.87	197	430	2.04	12.38	.33
.7	87.5	1.94	189	420	2.01	12.63	.33
.8	100.0	2.0	184	420	1.97	12.63	.33
.9	112.5	2.05	179	419	1.91	12.7	.32
1.0	125.0	2.1	175	415	1.88	12.8	.32

Assume each line power level = 20 milliwatts into 1000 ohms

Therefore E = 4.47 volts

$I = 4.47$ milliampere

$$E_{\text{induced total}} = E_1^2 + E_2^2 + \qquad + E_{1.0}^2$$

$$= .912 \text{ volts}$$

$$db = 20 \text{ LOG } \frac{4.47}{.912} =$$

$$= 13.8 \text{ db}$$

Thus it is seen that the crosstalk due to electrostatic induction is just at the threshold of intelligibility. All figures in the above computation are on a per 1000foot basis. It is to be noticed that length (ℓ) cancels out of the computation for E_{eq} since written in terms of ℓ

$$E_{eq} = \frac{E\,\ell C_1}{\ell c_1 + \ell c_2} = \frac{E\,C_1}{C_1 + C_2}$$

However, Zeq must be divided by ℓ since Zeq = $\frac{1}{w\,\ell(C_1 + C_2)}$. Therefore, to use the figures for longer lengths of wire Zeq must be divided by the number of 100' foot lengths of wire considered.

The above effects do not consider the coupling of energy due to image currents in the ground plane or conductive coupling. This effect is not computed here as it is an emperical computation at best. However, it should definitely be allowed for and with as close a db margin as shown: it is a factor in the choice between circuits. This effect will be considerable if the ground return is a wire and has several joints, for each joint may be considered as a boundary in the ground medium and transverse currents will be established, thus increasing conductive coupling.

Therefore, considering all the factors outlined above the choice in clearly in favor of a metallic return circuit over a single wire ground return circuit.

APPENDIX II

FREQUENCY PLAN

To convert the binary code in this scheme to frequency pulses a minimum of 7 frequency pulses or tones is required. It is desirable to space the tones across the frequency band of the voice channel and to so pick the frequencies that no tone frequency has a harmonic close to another frequency used as another tone.

A method of selecting frequencies[20] using geometric spacing may be used to select the desired frequencies which satisfies the above requirements. The set of frequencies is given by $f_n = a^{-1} f_1$

Here the constant percentage spacing is given by 100 (a-1). This series fits in nicely with use of tuned circuits having uniform value of Q and uniform frequency stabilities, properties easy to attain in practical circuits. This series also has the advantage that by carefully choosing the factor A it is possible to make the harmonics of each frequency fall approximately midway between higher adjacent frequencies in the same series. This is helpful in reducing interference from non-linear distortion.

Two frequency plans were used; one using a base frequency of 300 cycles, the other using a base frequency of 1000 cycles. In choosing a value for 'a' the factor of 1000 cycles must be considered, as this is the frequency used for ringing in this system. Thus in effect eight frequencies must be selected. The method for choosing A is shown in figures (23) and (24). The curves were computed by computing two points on each curve for two different values of as follows:

n	a^{n-1}	f_n	2 t_n	3 + n
1	1.0	300	600	900
2	1.1	330	660	990
3	1.21	363	726	1089
4	1.33	399	798	1197
5	1.46	438	876	1314
6	1.61	484	968	1452
7	1.77	533	1066	1599

$a = 1.5$

n	a^{n-1}	f_n	2 f_n	3 + n
1	1.0	300	600	900
2	1.1	450	900	1350
3	2.25	675	1350	2025
4	3.375	1010	2020	3030
5	5.06	1520	3040	4560
6	7.6	2280	4560	6840
7	11.4	3420	6840	

Let f = IKC

a = 1.2			a = 1.1		
n	a^{n-1}	f_n	n	a^{n-1}	f_n
1	1.0	1000	1	1.0	1000
2	1.2	1200	2	1.1	1100
3	1.441	1441	3	1.21	1210
4	1.716	1716	4	1.33	1330
5	2.15	2150	5	1.46	1460
6	2.49	2490	6	1.61	1610
7	2.985	2985	7	1.77	1770
8	3.582	3852	8	1.947	1947
9	4.3984	4398.4	9	2.1417	2141.7

The curves may be used to choose an optimum value of a since they display the relative frequency of each term in the series as a function of the spacing. Harmonics of f up the seventh are displayed on the right side of the graph. Points where a harmonic coincides with any frequency f_n are marked by dots and intervening points where a harmonic falls geometrically between two frequencies f_n and f_{n+1} are shown by circles. An ideal spacing factor A would be indicated by a vertical dotted line passing through a circle for every harmonic. Unfortunately the circle alignment is not this ideal, but by careful placement of the line a factor A may be chosen where the harmonics are fairly well interlaced. This is shown by the dotted lines in each figure. The frequencies chosen by these dotted lines are:

for $f = 300$ cycles
$a = 1.41$

n	a^{n-1}	f_n	$2 f_n$	$3 f_n$	$4 f_n$	$5 f_n$	$6 f_n$
1	1	300	600	900	1200	1500	1800
2	1.41	424	848	1272	1696	2120	2544
3	1.99	597	1094	1791	2388	2485	
4	2.805	841	1682	2523	3364		
5	3.955	1186	2372	3558			
6	5.56	1670	3340				
7	7.88	2360					

$a = 1.37$

n	a^{n-1}	f_n	$2 f_n$	$3 f_n$	$4 f_n$	$5 f_n$	$6 f_n$	$7 f_n$
1	1	300	600	900	1200	1500	1800	2100
2	1.37	411	822	1233	1644	2055	2466	
3	1.876	563	1126	1689	2252	2815		
4	2.57012	771	1542	2313	3084			
5	3.52106	1056	2112	3168				
6	4.82385	1447	2894					
7	6.60867	1983						

for $f = 1000$ cycles
$a = 1.135$

n	a^{n-1}	f_n	$2 f_n$	$3 f_n$
1	1.0	1000	2000	3000
2	1.135	1135	2770	3405
3	1.289	1289	2578	
4	1.462	1462	2924	
5	1.659	1659	3318	
6	1.882	1882		
7	2.14	2140		
8	2.43	2430		
9	2.76	2760		

The minimum spacing between frequencies is 130 cycles: the maximum spacing is 182 cycles.

Since the frequencies based on 1000 cycles present fewer problems in building filters, it is the group of frequencies chosen.

FREQUENCY SELECTION CURVE

FIG 23

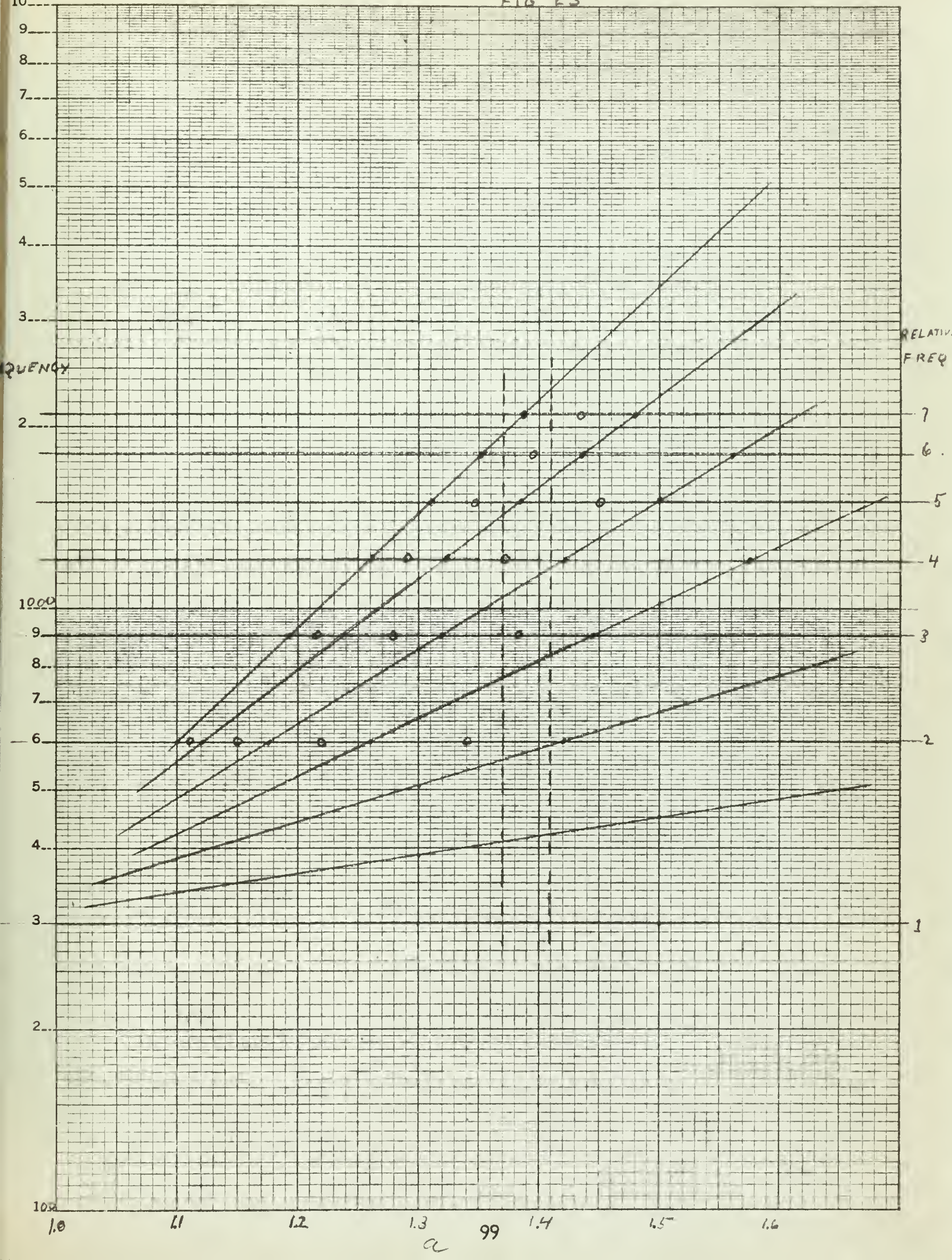


FIG 24

FREQENCY SELECTION CURVE